OBJECTIVE IDENTIFICATION OF HEART BEAT STATUS USING SPECTRAL ANALYSIS

M.c. THESIS

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Juanary 2021

HARAMAYA UNIVERSITY, HARAMAYA

Objective Identification of Heart Beat Status Using Spectral Analysis

A Thesis Submitted to the Collage of Computational and Natural Sciences
POSTGRADUATE PROGRAM DIRECTORATE

HARAMAYA UNIVERSITY

In Partial Fulfillment of the Requirements for the Degree of

MASTERS IN COMPUTATIONAL PHYSICS

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Juanary 2021

HARAMAYA UNIVERSITY, HARAMAYA

HARAMAYA UNIVERSITY

POSTGRADUATE PROGRAM DIRECTORATE

As Thesis Research Advisor, I hereby certify that I have read and evaluated this Thesis entitled **"Objective Identification of Heart Beat Status Using Spectral Analysis"** prepared under my guidance by Salih Mehamedyasin. I recommended that it be submitted as fulfilling the thesis requirement M.Sc. Degree in Computational Physics.

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As a member of the Board of Examiners of the M.Sc. Thesis Open Defense Examination, I certify that I have read and evaluated the Thesis prepared by Salih Mehamedyasin and examined the candidate. I recommended that the thesis be accepted as fulfilling the Thesis requirement for degree of Master of Science in Physics (Computational Physics).

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I dedicate this thesis manuscript to my families for their dedicated partnership for the success of my life.

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BIOGRAPHICAL SKETCH

The author was born in November 1989 in Abi-Adi town, Central Zone, Tigray region, Ethiopia. He attended formal education in Abi–Adi Primary School; Ras Alula Abanega Secondary and Preparatory School. After completion of his School education, he joined, Hawassa University in 2009 and graduated with BSc Degree in Physics July 02, 2011. Then after he was employed as a Physics teacher at Keranyo Medhani-Alem Secondary and Preparatory School at Addis Ababa City Educational Bureau. After serving for 4 years he joined Postgraduate Program of Haramaya University in July 2015 to pursue his M.Sc. study in Physics (Computational Physics) in the College of Natural and Computational Sciences.

ACKNOWLEDGEMENTS

First and foremost, I want to offer this endeavor to my Lord, ALLAH (SWT), for blessing me and also for giving me strength ,patience and protection throughout my postgraduate studies including the completion of this Thesis work.
I would like to express my sincere gratitude to my respected advisor Getachew Abebe (PhD) for giving me support and guidance , adding valuable comments, giving directions, suggestions and moral encouragement starting from the inception to the accomplishment of the study and also for his supervision in finishing this Thesis work.Without his encouragement, insight and professional expertise, the completion of this work would not have been possible.Most of all, I would like to mention about his beloved personality, extensive kindness towards students and unforgettable warm well come at this office.He was an energizer in my stay at the program.

I would like to thankful to the Ethiopian Ministry of Education (MoE) for financing the research work. I am also grateful to the Physics Department, College of Natural and Computational Sciences and Postgraduate Program Directorate of Haramaya University for hosting and rendering all rounded services during the study period.

My sincere thanks also go to Ashenafi G/slassie(MD) specialist in Abi-Adi General Hospital for giving the data or samples of normal and abnormal heart beats with classification in age groups for free.

My special thanks and heartfelt gratitude extends to my beloved family, specially my brothers Jemal and Aminu mehamedyasin for their encouragements.
Last but no means least, thanks go to my beloved wife Nejat mehamed for her words of encouragement and support as a source of strength throughout the course of my study.

ACRONYOMS AND ABBRIVIATIONS

AV	Atrioventricular Valve
BPM	Beats Per Minute
Db	Decibel
DFT	Discrete Fourier Transformation
FFT	Fast Fourier Transformation
FHS	Fundamental Heart Sound
FT	Fourier Transformation
LPF	Low Pass Filter
MATLAB	MATrix LABoratory
PCG	Phono CardioGram
PSD	Power Spectral Density
SV	Semi-lunar Valve

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ABSTRACT

Cardiac auscultation is one of the most important physical examination and a part of the first medical diagnostic procedures. In this study, the spectra of the heart beat sound to differentiate the normal and abnormal heart beat sounds of human with age groups have been examined. The sounds of the heart were captured using the stethoscope, mic of a personal computer, and then saved the recorded sounds in digital form. The heart beat sound data collected as time wave form, line spectra and PSDs of each of the sounds were plotted. The plot of line spectra and PSDs were done after the sound signals were transformed from the time domain to the frequency domain using FFT. The results of each of the plots showed that the sound signals due to the diastolic murmur and systolic murmur are different from in terms of the time waveform, the line spectrum and the PSD of the sound signal from the normal heart beat (Physiologically split). This is used to conclude that the sound signals due to diastolic murmur (mitral stenosis) and systolic murmur (mitral regurgitation) has specific line spectra as soon as the sound it has during that abnormal heart sound of human. The abnormal heart sound were obtained as being different from the normal heart beat of human.

Keyword: *FFT, Frequency domain, Heart, Line spectrum, PSD, Time domain, Time wave*

1. INTRODUCTION

1.1. Background of the Study

The stethoscope comes from the Greek language stethos means chest and scope means inspection. It is a very vital transducer for many medical practitioners including doctors, nurses and physicians to detect the abnormalities of the heart and lung such as sounds of heart, lung rhythm, and vibration of the intestines and blood flow (Geddes, 2005). The type of stethoscope used most of these days is the acoustic stethoscope (Myint and Dillard 2001).The problem with this acoustic stethoscope is the sound level it produce,which is very low(easily affected by the movement and noise of the surrounding) and thus makes it difficult to analyze and diagnose the heart sound. However, digital stethoscope can enhance the capability of acoustic stethoscope and can reduce the probleme.(Tay and Fancies ,2009; Jokic,2010*)*.

Cardiac auscultation is a technique of listening to heart sound as a result of any abnormality.The sound may indicate some problem in the heart and thus basic analysis tool must be used to evaluate the function of the heart (Prakash *et al.*, 2013).Usually, stethoscope helps to listen to heart sound.Heart rate is the speed of the heart beat measured interms of the number of heart beat per unit time expressed as beat per minute (bpm). The heart rate of a healthy adult at rest is around 72bpm & babies at around 130bpm, normal resting heart rates range from 60-100 bpm, while older children have heart beat rates of around 90bpm.The heart rate rises gradually during exercises and returns slowly to its normal rate after exercise (Bauer, 2008).

The heart sound comprises of four components: S_1, S_2, S_3 and S_4. S_1 (known as “lub”) and S_2 is (known as “dub”) are the fundamental heart sound (FHS). S_1 is caused due to closure of atrioventricular valves.The region between S_1 and starting of S_2 of the same heart cycle is systolic region. Systolic refers to the pressure of blood in the artery when the heart contracts.S_2 is caused due to closure of semi lunar valves. The region between S_2 and starting of S_1 of next heart sound cycle is diastolic. Diastolic refers to the pressure of blood in the artery when the heart relaxes between beats. S_3(proto diastolic or ventricular gallop) is produced when blood rushes into an incompletely emptied ventricle during rapid ventricular filling,while S_4(presystolic or atrial gallop) is produced by the sound of blood being forced into a stiff or hypertrophic ventricle.S_3 and S_4 are rare heart sounds, which are not normally audible.

Generally , heart sounds are classified as normal heart sounds and murmurs. Heart murmur or noise is the sound lower than the amplitudes of S_1 or S_2 and it appears occasionally in the cardiac cycle due to the effect of diseases or age (Mehta and Khan, 2004).Normal heart sound is a result of opening and closing of heart valves that consists of two sounds $S_1$and S_2 which occur periodically in sequence with every heart beat. The normal heart sound components are described as follow (Samruddhi *et al.*, 2017).

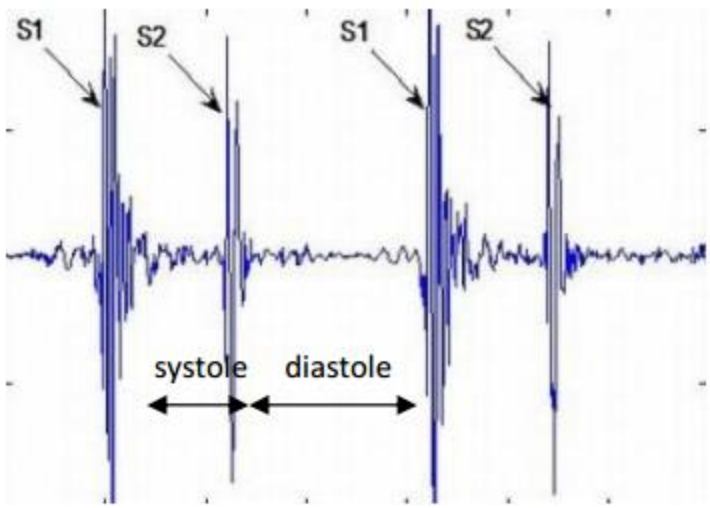

Figure 1: **Normal Heart sound (Samruddhi *et al.*, 2017).**

Heart murmurs are produced as a result of turbulent flow of blood. Turbulence is the whooshing sounds resulting due to turbulent flow of blood through the heart valves that is suffiecient to produce audible noise.The blood turbulence is mainly caused due to opening and closing of heart valves and also due to fast acceleration and retardation of blood flow in the hear chamber (Yan , 2010a). Most of the murmurs are not audible in the auscultation.Figure 1.2 shows the phonocardiograms on auscultation of from normal and abnormal heart sounds. PCG is digital recording of heart sound.

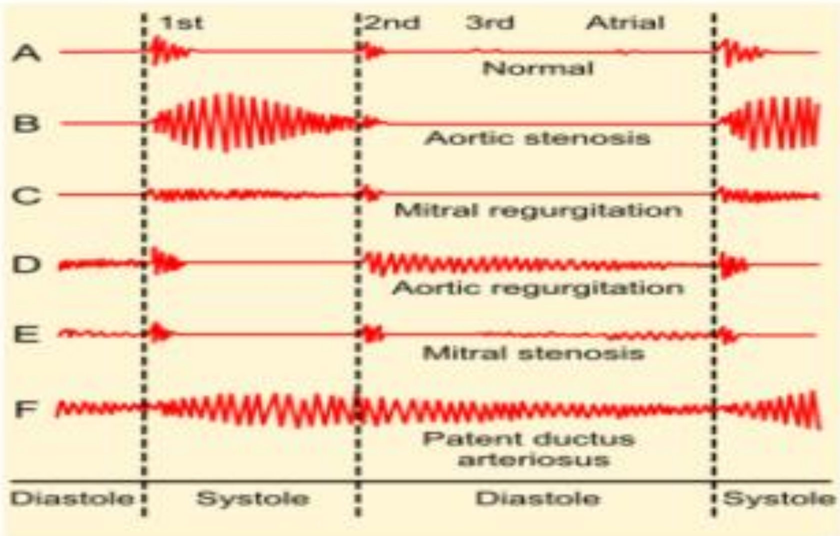

Figure 2: Normal and abnormal heart sounds (Yan , 2010a).

- Regurgitation through the mitral valve is by far the most commonly heard murmur, sometimes fairly loud to an experienced ear , even though the volume of regurgitate blood flow may be quite small.
- Stenosis of the aortic valve is typically the next most common heart murmur, a systolic ejection murmur. This is more common in older adults or in those individuals having a two, not a three leaflet aortic valve.
- Regurgitation through the aortic valve, if marked, is sometimes audible to an experienced ear with a high quality, especially electronically amplified, stethoscope. Generally, this is a very rarely heard murmur, even though aortic valve regurgitation is not so rare. Aortic regurgitation, though obvious using echocardiography visualization, usually does not produce an audible murmur.
- Stenosis of the mitral valve, if severe, also rarely produces an audible, low frequency soft rumbling murmur, best recognized by an experienced ear using a high quality, especially electronically amplified, stethoscope.
- Either regurgitation through, or stenosis of, the tricuspid or pulmonary valves essentially never produces audible murmurs.
- Audible murmurs are associated with abnormal openings between the left ventricle and right heart or from the aortic or pulmonary arteries back into a lower pressure heart chamber.

Table 1. Murmurs Description(Mohamad , 2015)

Graduation of murmur	Description (defined based on use of an acoustic,not a high –fidelity amplified electronic stethoscope)
Grade1	Very faint,heard only after listener has “tuned in ‘;may be heartd in all positions
Grade2	Quiet,but heard immediately after placing the stethoscope on the chest
Grade 3	Moderately loud
Grade 4	Loud,with palpable thrill (i.e a tremor or vibration felt on palpation)
Grade 5	Very loud,with thrill.May be heard when stethoscope is partly off the chest
Grade 6	Very loud,with thrill. May be heard when stethoscope entirely off the chest

1.2. Statement of the Problem

Detection and analysis of heart beat is very difficult to medical doctors and physicians due to the very low signal level. The symptoms of hearts beat are varying depending on the severity of the condition. Symptoms can include: the sensation of having a 'thumping heart' (palpitations). But note: sensations of palpitations are also common in people who do not have an arrhythmia.

The heart sound diagnoses were undertaken through subjective interpretation.It usually has the following problems due to the fact that :-

- ✓ Both heart sounds and murmurs are relatively low frequency ranging between 10 to 1000 Hz making them less audible to be heard clearly by the physician.
- ✓ The audibility varies from person to person making the auscultation process dependent on the skills and expertise of the physician.
- ✓ Heart murmurs are different by certain sound characteristics: loudness or intensity, frequency, quality, timing (systole and diastole), shapes of the murmur

and radiation due to sex, age. Radiation is one of the heart murmur characteristic reflects the intensity of the murmur and the direction of blood flow.

- ✓ Many pathological condition of the cardiovascular system causes murmur and aberration in heart sound much before they are reflected as other symptoms.

Thus, to overcome these difficulties, this study uses the techniques of spectral analysis to objectively identify the normal (physiologically split) and of abnormal (heart murmur) heart status. The main interest of this work is to develop computer routine algorithm (using MatLab software) that can identify the normal and abnormal heart sound (heart murmur) produced due to normal and abnormal status of heart respectively.

1.3. Objectives of the Study

1.3.1. General objective

The general objective of this study was to develop sound based techniques that can identify heart beat status and to examine the pattern of heart beat using spectral analysis.

1.3.2. Specific objectives

- ➢ Determine the pattern of the human normal heart beat for 0-15 age group.
- ➢ Select 0-15 age group level of normal heart beat(physiologically split) of human to get abnormal heart beat from the human patient and compare the human normal heart beat and abnormal heart beat (diastolic(mitral stenosis) and systolic(mitral regurgitation) murmur with same age group
- ➢ Identify the frequency at which the dominant power spectrum in the frequency domain occurs for the same age group.

1.4. Significance of the Study

The purpose of this study was to develop computer routine algorithm to identify the abnormal heart sound (heart murmur) using spectral analysis.The spectral representation of the sound produced when different heart beat abnormalities is highly essential for the identification of the heart condition using sound signal processing.

A sound signal can be sampled to result in a discrete (individually separate and distinct) signal (Attaway, 2009 and Mcloughlin, 2009). To determine whether the heart beat is normal or abnormal, an important diagnostic tool of hearts' sounds produced by various mechanical activities of the heart during the heart cycle can be used (Prema *et al.*, 2010).This study investigates the task of computer aided diagnosis which can be employed in the field of medicine. The heart beat signal was used as an input to the developed signal processing software. The software employees spectral analysis techniques to identify the heart sound signal obtained using the electronic stethoscope.

2. REVIEW OF LITERATURE

2.1. Heart

The heart is a powerful muscle, about the size of a clenched fist, located at the left of the centre of the chest with the special type of muscle called the myocardium.The heart functions as a rhythmic, automatically repeated beating maintained by electrical impulse originating in the site to supply the blood and oxygen to all parts of the body.The heart is located in the chest cavity just posterior to the breastbone, between the lungs and superior to the diaphragm. Blood is pumped away from the heart through arteries and returns to the heart through veins. The major artery of the body is the aorta and the major veins of the body are the venacava (Sepehri, 2010). Figure 3 shows the heart of human.

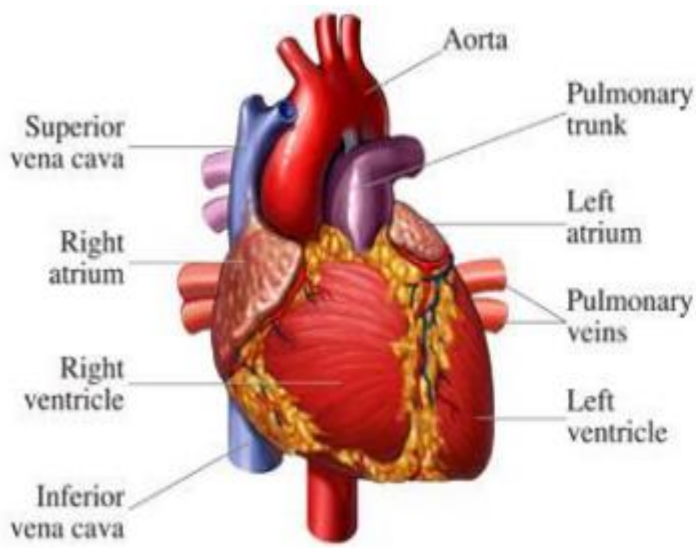

Figure 3 : Human's Heart (Sepehri, 2010)

2.1.1. Heart valves

The heart contains 4 valves (2 on the right side and 2 on the left side) that open and close to direct proper blood flow through the heart and to the rest of the body.

Four types of heart valves are:

- Tricuspid valve: allows flow from the right atrium to the right ventricle.

- Pulmonary valve: from the right ventricle to the pulmonary artery.
- Mitral valve: from the left atrium to the left ventricle.
- Aortic valve: from the left ventricle to the aorta.

2.2. Heart Sounds

The heart sounds are the noises generated by the beating heart and the resultant flow of blood through artery (the turbulence created when the heart valves snap shut).In cardiac auscultation, an examiner uses a stethoscope to listen to these sounds, which provide important information about the condition of the heart (Yan , 2010b).

In healthy adults, there are two normal heart sounds often described as a lub and a dub or dup, that occur in sequence with each heart beat. These are the first heart sound (S_1) and second heart sound (S_2), produced by the closing of the atrial ventricular valves and semi-lunar valves respectively (Amit *et al.*,2009).The semi lunar valves (SV) are the two valve structures that sit between the right ventricle and the pulmonary artery and between the left ventricle and the aorta. The atrioventricular valves (AV) allow blood to pass from the atria to the ventricles, closing tight to block leakage of blood back into the atria.

2.2.1. Murmur

The term murmur only refers to a sound believed to originate with in blood flow through or near the heart; rapid blood velocity is necessary to produce a murmur. Yet most heart problems do not produce any murmur and most valve problems also do not produce an audible murmur (Mohamad, 2015).Not every murmur is associated with valve disease. It can also be caused by conditions that may temporarily increase blood flow such as: exercise, pregnancy, fever, hyperthyroidism, anemia, rapid growth spurts(forcible gush; sudden outburst; stream, jet, squirt; marked increase of effort for a short time)in children.

In this study the digital recording of the heart sound was studied and classified into three classes namely normal heart beat, diastolic murmur and systolic murmur of 0-15 age group. Sound signal amplified using the electronic stethoscope to identify heart beat status using spectral analysis.

A. Systolic murmur

The systolic murmur may be caused due to stenosis of semi lunar valve and regurgitation of the atrioventricular. Most commonly found between sounds S_1 and S_2. These can be innocent or pathologic. Systolic murmur is derived from increased turbulence associated with increased flow across normal semilunar valve or into a dilated great vessel, flow across an abnormal semi lunar valve or narrowed ventricular outflow tract - e.g. aortic stenosis, flow across an incompetent atrioventricular valves(AV). e.g. mitral regurgitation and flow across the interventricular septum. Systolic murmurs may be early systolic, midsystolic, late systolic, or holosystolic.

B. Diastolic murmur

The diastolic murmur begins with or after S_2 sounds or before S_1 sounds. All diastolic murmurs are pathologic. Three main causes are known for diastolic murmur: aortic or pulmonary valve incompetence, mitral or tricuspid stenosis, and increased blood flow across mitral or tricuspid valves.

2.3. Stethoscope

The stethoscope was invented in 1816 when a young French physician named ReneTheophileHyacinthe Laennec was examining a young female patient. It is often used to listen lung and heart sounds.

Table 2.The components of Stethoscope shown in Figure 4.

Label	Name
1	headset
2	Ear tip
3	Ear tube
4	Tunable Diaphragm
5	Stem
6	Tubing
7	Chest piece

The chest piece is the part of the stethoscope that is placed on the location where the user wants to hear sound (Mohamed, 2015).

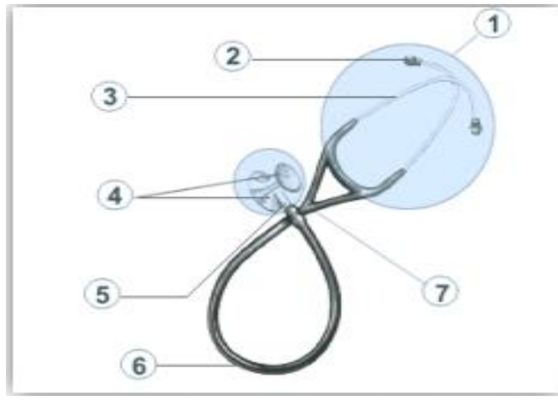

Figure 4 : Stethoscope part (Mohamad ,2015)

2.4. Related Works

Heart sounds are the acoustic waves generated by the beating heart and resultant flow of blood through it. It is a mechanical vibration caused by the myocardial systolic/diastolic, valve opening and closing, blood flow impact in ventricular wall and large artery in the heartbeat cycle. The entire signal is contained in a wide range of low frequencies, described as a 5-600Hz (Wang et al., 2009) or 50-500Hz (Yuel et al.,2012). Above 120Hz is usually considered as high-frequency sound, 80Hz-120Hz as a middle-frequency sound, below 80Hz as a lowfrequency one. From physiologically point of view their occurrence is normal. First heart sound (S1) is occurs as a result from reverberation within the blood associated with the sudden block of flow reversal by the valves. This bass tone is about 40-60Hz frequency and continues to about 0.14-0.16s (Song et al., 2012). Second heart sound (S2) is a diastolic tone, because it occurs during the diastole of the ventricular muscle.The second heart tone is a vibration of frequency of 60-100Hz and continues to about 0.08-0.12s. The heart sounds other parameters, such intervals between S1 and S2, between S2 and S1, heart sound gap period, have an important value. Their duration varies with age and might be a symptom of a disease. The average heartbeat cycle of adult is about 0.8s, which includes 0.3s systole and 0.5s diastole.

2.5. Fourier Transform

2.5.1. Fourier series

Fourier series provides an alternative way of representing data; instead of representing the signal amplitude as a function of time, one can represent signal by how much information is contained at different frequencies (Attia, 1999; Boggess and Narcowich, 2009; Mandal Asif, 2007). A Fourier series takes a signal and decomposes it in to a sum of sine and cosine of different frequencies (Jeffrey, 2002; Karris, 2004). Assume that we have a sound signal that can be written in the form of equation. One can represent that signal by the infinite series

$$f(t) = \frac{a_0}{2} + \sum_{n=1}^{\infty}\left(a_n \cos(n\omega t) + b_n \sin(n\omega t)\right) \qquad (2.1)$$

Where $\omega = \frac{2\pi}{T}$, f (t) is the signal in the time domain, and a_n and b_n are the unknown coefficients of the series. For a signal of period *2π*, these coefficients are:

$$\begin{cases} \frac{a_0}{2} = \frac{1}{2\pi}\int_0^{2\pi} f(t)dt \\ a_n = \frac{1}{\pi}\int_0^{2\pi} f(t)\cos(n\omega t)dt \\ b_n = \frac{1}{\pi}\int_0^{2\pi} f(t)\sin(n\omega t)dt \end{cases} \qquad (2.2)$$

There are different interesting forms of Fourier series (Attia, 1999; Karris, 2004). For example, the expression in equation (2.1) is the trigonometric form of Fourier series (Karris, 2003). One of the other forms is known as the harmonic form of Fourier series (Hsu, 1995) that is written as:

$$f(t) = \frac{a_0}{2} + \sum_{n=1}^{\infty}(A_n \cos(n\omega t + \theta_n)); \qquad (2.3)$$

Where

$$\begin{cases} A_n = \sqrt{a_n^2 + b_n^2} \\ \theta_n = -\tan^{-1}(\frac{b_n}{a_n}) \end{cases} \qquad (2.4)$$

Another form of Fourier series can be given by the exponential equivalent that is obtained through Euler's formula (Jeffrey, 2002; Karris, 2004**):**

$$f(t) = \sum_{n=-\infty}^{\infty} c_n\, e^{jn\omega t} \qquad (2.5)$$

Where

$$c_n = \frac{1}{2\pi}\int_0^{2\pi} f(t)e^{-jn\omega t}\,dt \qquad (2.6)$$

2.5.2. Discrete Fourier transform (DFT)

Discrete Fourier transform (DFT) is the Fourier transform adapted for digital signal processing. The transformation of discrete data between the time domain and frequency domain is quite useful in extracting information from the signal. The DFT expresses signals as a linear combination of sinusoidal or complex exponential; signals with various angular frequencies (Attia, 1999; Elali, 2005; Orfanidis, 2010). This decomposition of signals allows one to examine the effects of the system on each signal component.

DFT plays a central role in the implementation of many signal processing algorithms. It is a mathematical transform which resolves a time series x[n] into the sum of an average component and a series of sinusoids with different amplitudes and frequencies (Musoko, 2005). To compute the frequency content of a signal (or the frequency response of a system) we use the DFT (Mandal and Asif, 2007; Orfanidis, 2010; Rocchesso, 2003):

$$X[\omega] = \sum_{n=-\infty}^{\infty} x[n]e^{-jn\omega} \qquad (2.7)$$

The fact that x[n] is a finite length sequence implies that the DFT can be rewritten as

$$X[\omega] = \sum_{n=0}^{N-1} x[n]e^{-jn\omega} \qquad (2.8)$$

Computing the DFT for only a finite number of frequency points means that we can further simplify equation (2.8) to

$$X[\omega] = \sum_{n=0}^{N-1} x[n]e^{-jn\omega k} \qquad (2.9)$$

Where $\omega_k = \frac{2\pi k}{N}$: are the frequency sample: if we assume that there are N samples, then k = 0,1,2,…,N-1. That is, we are evaluating $X[\omega]$ only at the frequency values $\omega_k = \frac{2\pi k}{N}$ for k=0,1,2,…,N-1. The resulting expression is:

$$X[\omega] = \sum_{n=0}^{N-1} x[n]e^{-j\frac{2\pi kn}{N}} \qquad (2.10)$$

The N-point Discrete Fourier Transform (DFT), X[k], of an N-point discrete-time sequence, x[n], is defined as:

$$X[k]=\sum_{n=0}^{N-1} x[n]e^{-j\frac{2\pi kn}{N}} \qquad (2.11)$$

For k = 0, 1, 2, …, N-1. The DFT, X[k], is just a sampled version of the DFT, X $[\omega]$, at $\omega = \frac{2\pi k}{N}$.We can write this equation as:

$$X[k]=\sum_{n=0}^{N-1} x[n]\, W_N^{kn} \qquad (2.12)$$

Where

$$W_N = e^{-j\frac{2\pi}{N}} \qquad (2.13)$$

The factor W, also known as the twiddle factor, is a function of N frequency terms with argument kn which can take on integer value up to $(N-1)^2$(Hsu, 1995; Wong, 2006). Each point of the DFT in equation (2.12) can be calculated using N complex multiplications and (N-1) complex additions. Therefore, N^2complex multiplications and N (N-1) complex additions are needed to compute NDFT (Orfanidis, 2010).

We may express the N-point DFT in matrix form (Orfanidis, 2010; Rocchesso, 2003) as:

$$X_N = W_N x_N \qquad (2.14)$$

Where W_N the N x N matrix of linear transformation is x_N is the N-point vector of the signal x[n], and X_N is the N-point vector of frequency samples. So, the above equation (2.14) can be denoted in matrix form as shown below

$$\begin{bmatrix} X(0) \\ X(1) \\ X(2) \\ \vdots \\ X(N-1) \end{bmatrix} = \begin{bmatrix} 1 & 1 & 1 & \cdots & 1 \\ 1 & W_N & W_N^2 & \cdots & W_N^{N-1} \\ 1 & W_N^2 & W_N^4 & \cdots & W_N^{2(N-1)} \\ \vdots & \vdots & \vdots & \vdots & \vdots \\ 1 & W_N^{N-1} & W_N^{2(N-1)} & \cdots & W_N^{(N-1)2} \end{bmatrix} \begin{bmatrix} X(0) \\ X(1) \\ X(2) \\ \vdots \\ X(N-1) \end{bmatrix} \qquad (2.15)$$

The process of matrix multiplication requires N multiplications for its completion. If large amounts of data are to be processed, this can become inordinate, even for a computer. The secret of the fast Fourier transform is that, it reduces the number of multiplications to be done from N^2 to about 2N $\log_2$(N)(James,2011).

2.5.3. Fast Fourier transforms (FFT)

Discrete Fourier transform would normally require O(N^2)time to process for n samples: There are many types of FFT algorithms available. One of them, which is most commonly used is the radix-2 algorithm. This algorithm was developed by (Cooley and Tukey, 1965). It assumes the data set N is a power of 2 and hence is called a radix-2 algorithm.

If a sequence x[n] is separated into two point sequences with one corresponding to the even numbered and the second representing odd numbered samples of x[n], the equation (2.11) becomes (Mulgrew *et al.*, 2003):

$$\mathrm{X}[k] = \sum_{n=0}^{\frac{N}{2}-1} x\,[2n] W_N^{2kn} + \sum_{n=0}^{\frac{N}{2}-1} x[2n+1] W_N^{2(2n+1)} \qquad (2.16)$$

2.6. Sound Signal Visualization

MatLab as used to analyze, reconstruct and play the sound. Several aspects of these procedures are controlled and rated for their importance, i.e. how they affect the recognizable features of a sound. Since sound is an actual wave its amplitudes are stored in the sound signal.The plot function was used to display the sound data recorded.The different plots of the data were analyzed as follows. Three sound signal visualization techniques were used to analyze the sound signal graphically.

2. 6.1. Time waveform

This is one of the tools of sound signal visualization techniques through which the sound data records can be displayed and useful, especially for a quick scan of very long recordings(Mcloughlin, 2009).

2. 6.2. Line spectrum

Spectrum is a function we used frequently in MatLab to look at the spectral envelope of a recorded sound data (Mcloughlin,2009; Pozrikidis, 2005).Spectrum takes in a vector of numbers from a".wav" file and the sampling frequency of the waveform.

The fast fourier transform (FFT) is probably the most popular transform used to obtain the frequency spectrum of a signal. When the FFT coefficients are known, it is useful to plot the amplitudes of the harmonics on a frequency domain showing the first (fundamental frequency) harmonic, and the higher harmonics times the amplitude of the fundamental (Stoica and Moses, 1997). Analyzing the sound spectrum means finding out how many different frequencies and amplitudes make up a sound at a particular moment.

2.6.3. Power spectral density (PSD)

Another quantity commonly used in sound analysis technique is the power spectrum or spectral density. The absolute value of the FFT coefficients will provide one with the total amount of information contained at a given frequency, the square of the absolute value is considered the power of the signal. It is simply defined as the square of the Fourier magnitude:

$$\mathrm{P(k)} = |X[k]|^2 \qquad (2.17)$$

A plot of the power spectrum of a signal gives us an idea of the density of power at different frequencies (Mcloughlin, 2009; and Pozrikidis, 2005).

The loudness level of the sound signal that is related to the power contained in the signal can be expressed as:

$$\mathrm{L\ (dB)} = 10\times \log 10\left(\frac{p(k)}{p(r)}\right) \qquad (2.18)$$

Where L (dB) loudness level in decibel, $p_{(k)}$ is the measured power of the signal and $p_{(r)}$ is the reference power of the signal (in put power signal).

A decibel (abbreviated dB) is defined as one tenth of a bel; which is a unit for measuring the loudness of a sound. Since sound signal intensity, sound signal power, and sound signal energy are always proportional to the square of the signal amplitude. Thus, we can always translate these energy related measures into squared amplitude; (Mcloughlin, 2009; and Pozrikidis, 2005). One can also express equation (2.17) in terms of the sound signal amplitude as:

$$\text{Amplitude in dB} = 20 \times \log_{10}\left(\frac{signal\ Amplitude}{reference\ Amplitude}\right) \qquad (2.19)$$

3. MATERIALS AND METHODS

3.1. Description of the Study

The objective of this study was to identify the normal heart sounds (physiologically split) and abnormal heart sounds (diastolic (mitral stenosis) and systolic (mitral regurgitation) murmurs of 0-15 age group using spectral analysis. In this work, the sound produced due to normal heart beat and abnormal heart sounds were recorded using computer with MatLab soft-ware. The heart sound signals were obtained using a stethoscope which was transferred to computer for further analysis.

3.2. Experimental Apparatus

To perform this study, stethoscope, and personal computer were used .In the process, first the sounds of the normal heart beat 0-15 age were recorded.The sound produced as a result of abnormal heart sound (diastolic and systolic) murmur could be compared with the sound recorded from normal heart beat of the respective age group.

3.3. Experimental Setup

The capturing of sound from normal heart beat and abnormal heart sound (diastolic and systolic) murmur was made using the apparatuses arranged as shown in the Figure 5.The recordings of the heart sound were connected with stethoscope undertaken in an isolated room.The stethoscope also attached with the mike of the personal computer. Sound signal preprocessing using low pass filter (LPF) was used to minimize the noise.

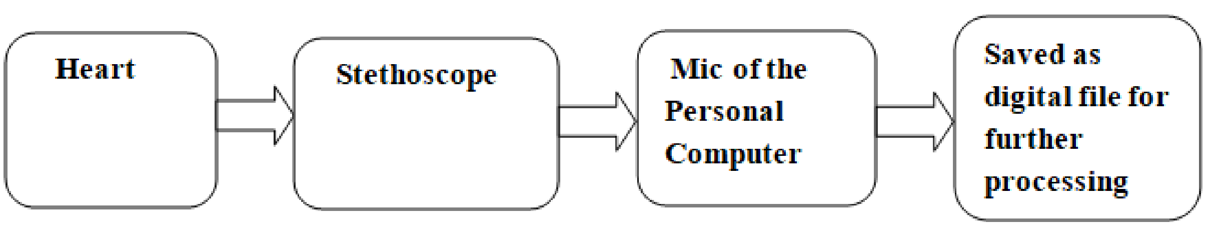

Figure 5: Experimental set up for heart sound

MatLab function used in the record of the heart sound (normal and abnormal) was;

Y = wavrecord (Ns, Fs, Nc, ‟ dtype ‟) (Alasdair, 2004).
Where

- Ns are the total number of data samples to be recorded,
- Fs are the sample frequencies (samples per second),
- Nc are the number of input channels used to record the data which is 1 for mono and 2 for stereo,
- dtype are the data type to be recorded, specified by string i. e., “single” , “double”, “int8”, “int16”.

3.4. Capturing Sound Samples

3.4.1. The procedure of the study

The heart beat sounds of normal heart and abnormal heart sounds were captured 0-15 age groupe level. For this study, nine sampled of heart sound sources.They were three sounds of normal heart beat (physiologically split) of human, three sound signal of diastolic murmur (mitral stenosis), and three sound signal of systolic murmur (mitral regurgitation) for the age group ranging from 0-15. The heart beat sounds were recorded with the assistance of heart specialist/cardiologist. Three types of plots i.e. time wave form; line spectrum and power spectral density were investigated. When carrying out experimental analysis, the goal was to obtain good result by using the appropriate stethoscope: sampling frequency and number of samples.The sound signal of heart beat in both normal and abnormal heart beat condition with 20 kHz sampling frequency and 131072(2^{17}) number of samples were recorded in a wave format. The input signals were a real-time sound signals from stethoscope.The sample frequency is the number of samples recorded per second during sound recording process and the time taken to record this sound data length was obtained to be 6.55seconds, which is :

$$T_s = \frac{N_s}{F_s} = 131072/20000 = 6.55 \text{ seconds}$$

3.5. Method of Data Analysis

3.5.1. Sound signal analysis

Three plots for visualization of heart beat sounds are being used. These are Time wave, Line spectrum and Power spectral density(PSD).

A. Time-wave

The time wave form representation of heart signal conveys little information about frequency content thus the sound signal has to be transformed in to frequency domain. However for immediate visualization of the signal particularly the intensity and the wave pattern, the analysis of sound signal the time wave plot could be used. Extract the sound signal using time wave form of sound from normal heart beat, diastolic murmur and systolic murmur.

B. Line spectrum

Fast Fourier transformation is employed to obtain the frequency spectrum of signal.The analysis of the sound frequency spectrum envisaged the amplitude constituents at different frequencies. The heart sound of normal heart beat (physiologically split),diastolic murmur(mitral stenosis) and systolic murmur(mitral regurgitation) were visualized in the frequency domain. Such representation was highly essentially to clearly extract the hidden information contained in the signal and the peak value (amplitude level) of sound signal for normal heart beat and abnormal heart sound (systolic and diastolic murmur). The line plot could be depicted as the amplitude level against frequency from which the dominant frequency could be determined.

C. Power spectral density (PSD)

Another quantity commonly used in sound analysis technique is the power spectrum or spectral density. PSD is a sound analyzing technique that shows how the density of power of the sound signal is spread over the frequency. It establishes the amount of power on each frequency. The power spectral density of sound from normal heart beat, diastolic and systolic murmur was extracted .The extracted PSD shows the plot of the intensity sound signal which has the maximum value for normal heart beat and abnormal heart sound. The PSD results of the normal and abnormal heart beat (diastolic and systolic murmur) for 0-15 age group were compared with each other.

4. RESULTS AND DISCUSSIONS

For this study, three types of sound sources and each sound signal were three samples. They were three sounds of normal heart beat (physiologically split) of human, three sound signal of diastolic murmur (mitral stenosis), and three sound signal of systolic murmur (mitral regurgitation) for the age group ranging from 0-15.

4.1. Sound of Normal Heart Beat

Three sound of normal heart beat of human recorded.

Table 3. Samples taken for nomal heart beat.

Sample	Age
n_1	11
n_2	5
n_3	2

4.1.1. Normal heart beat of sample n_1

4.1.1.1. Time waveform of sample n_1

The time waveform representation of the normal heart sound signal is very essential although it conveys little information about that signal of normal heart of sample n_1 of human recorded for 11 age group.The MatLab code in appendix A is used to read the recorded sound for normal heart of sample n_1 of human filters using the low pass filter (LPF) and plot the logarithm of the intensity versus time .

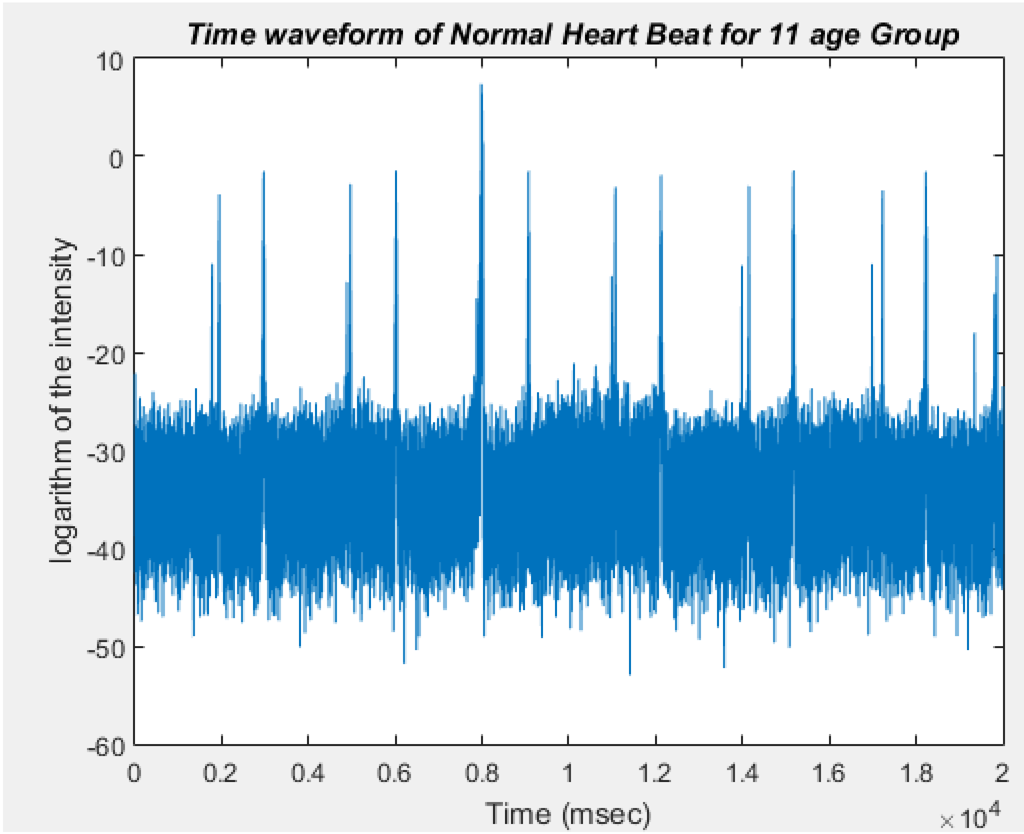

Figure 6 : Time waveform for normal heart of sample n_1

Figure 6 shows the plot of sound signal as a function of time. Normally we get and visualize a signal in time domain meaning that the horizontal axis is time and vertical axis is logarithm of intensity for normal heart beat of human and the child aged 11. There are different peak values correspendes to ‘‘lub’’ and ‘‘dub’’ of normal heart sound 1st heart sound, S_1 (lub), marks the beginning and end of systole precedes carotid pulse. It have 7 smaller gaps between peak values of the systole and an average of the smaller gaps of systole 0.12 beat per msec. The 2nd hear sound, S_2 (dub), marks the end of systole (beginning of diastole) follows carotid pulse. It have 6 higher gaps between peak values for diastole and an average of the higher gaps 0.2 beat per msec. From the review of literature the normal heart beat per minute on average 130bpm. On this thesis the normal heart beat on average beat per minuts 153bpm. The frequency information of the sound signal is hidden because of the fact that the time waveform reveals the sound signal as a function of time. However, the frequency information of the signal is vital to learn what constitute the signal and at what level. This may be achieved using FFT.

4.1.1.2. Line spectrum of sample n_1

Line spectrum is a sound analysis technique that plots of amplitude versus frequency of the sound signal which is converted from time to frequency domain. This conversion is done using FFT. The MatLab code that transforms the time domain signal to frequency domain, filters the sound from unwanted additional sound and plots its line spectrum is given in appendix A.

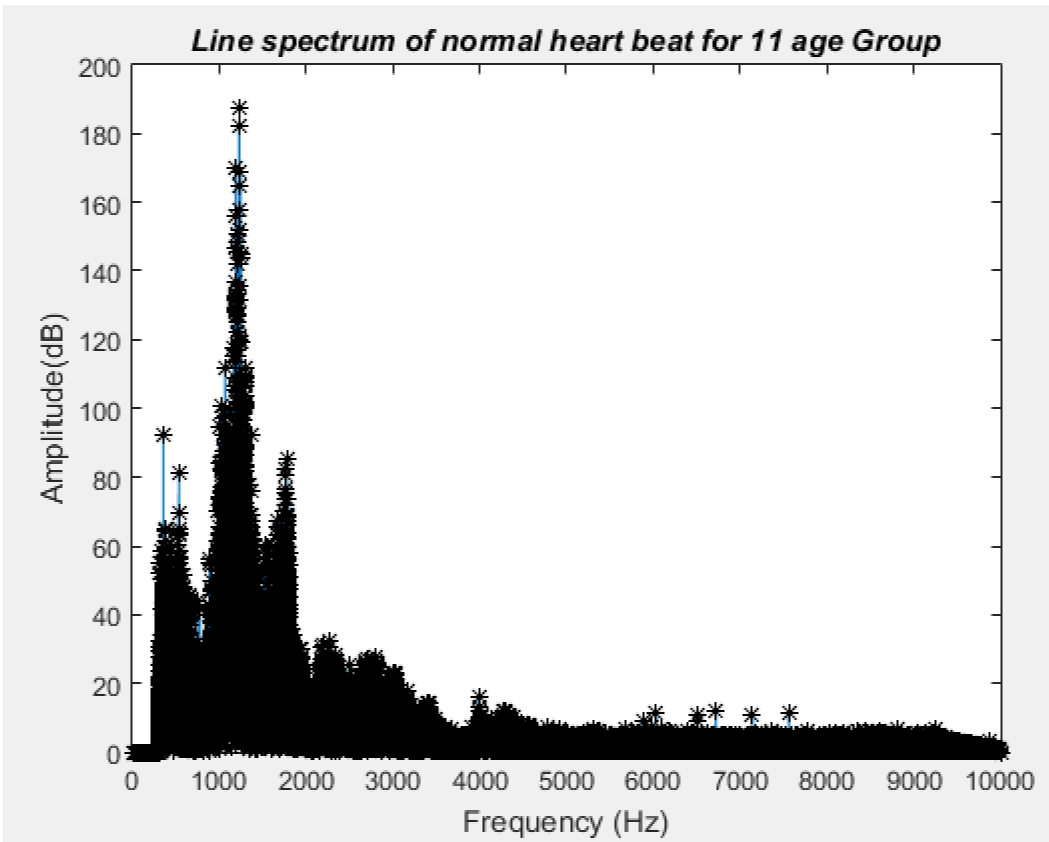

Figure 7 : Line spectrum for normal heart of sample n_1

Figure 7 shows the line spectrum of the sound signal of normal heart beat of human for 11 age group. The amplitude of the sound signal as a function of frequency. This plot also shows that the frequency is 1200 Hz and the maximum amplitude exhibited a level which lies 190dB.

4.1.1.3. PSD of sound of sample n_1

This technique of sound analysis is also very essential since it shows how the density of the power of the sound signal is spread over the frequency. This possibility is obtained after the conversion of the sound signal from the time domain to the frequency domain. This conversion is performed by the MatLab code written in appendix A which also plots the PSD of sound for normal heart beat of sample n_1 of patient of 11 years old.

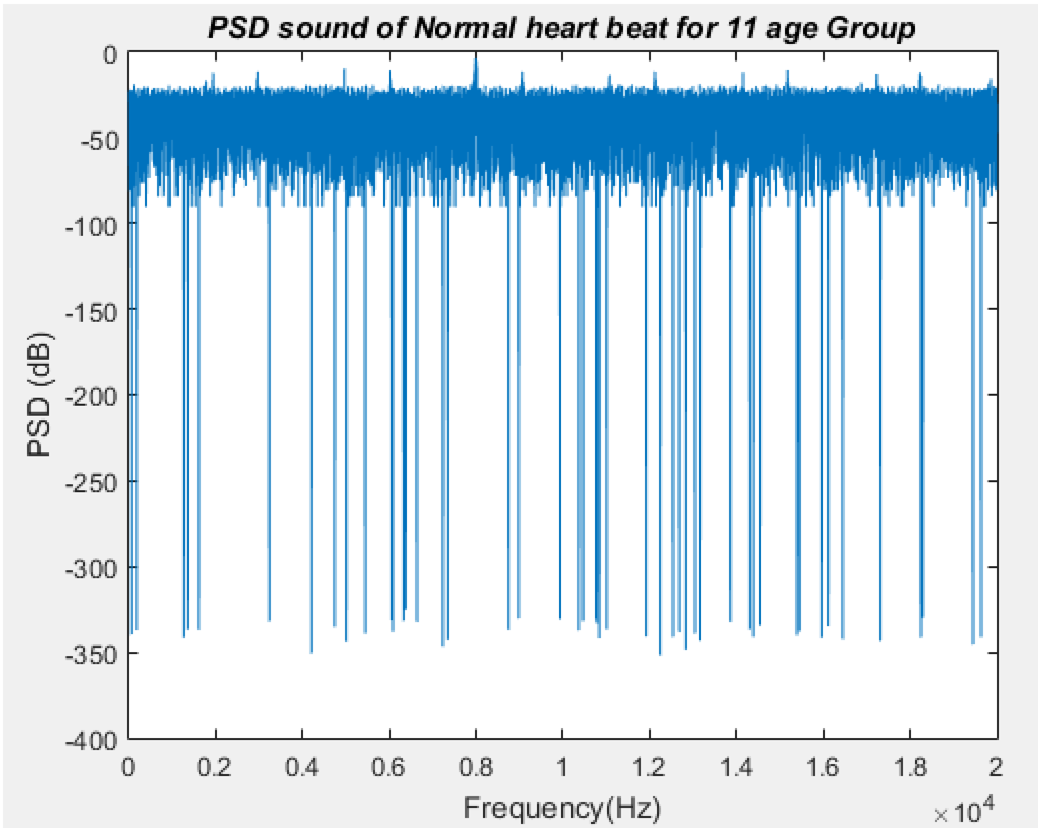

Figure 8: PSD of sound for normal heart beat of sample n_1

Figure 8 the plot of PSD of sound signal as a function of frequency.The plot clearly shows the distribution of the PSD of sound signal over the frequencies. The plot is for the sound from the normal heart beat of sample n_1 of patient of 11 years old. The PSD of the sound signal approximately -350dB which is acquired at a dominant frequency value approximately is 12KHz.

4.1.2. Normal heart beat of sample n_2

4.1.2.1. Time waveform of sample n_2

The time waveform representation of the normal heart beat of sample n_2 of human recorded of 5years old patient . The MatLab code in appendix B is used to read the recorded sound for normal heart beat of sample n_2 and plot the logarithm of the intensity versus time as shown in Figure 9 .

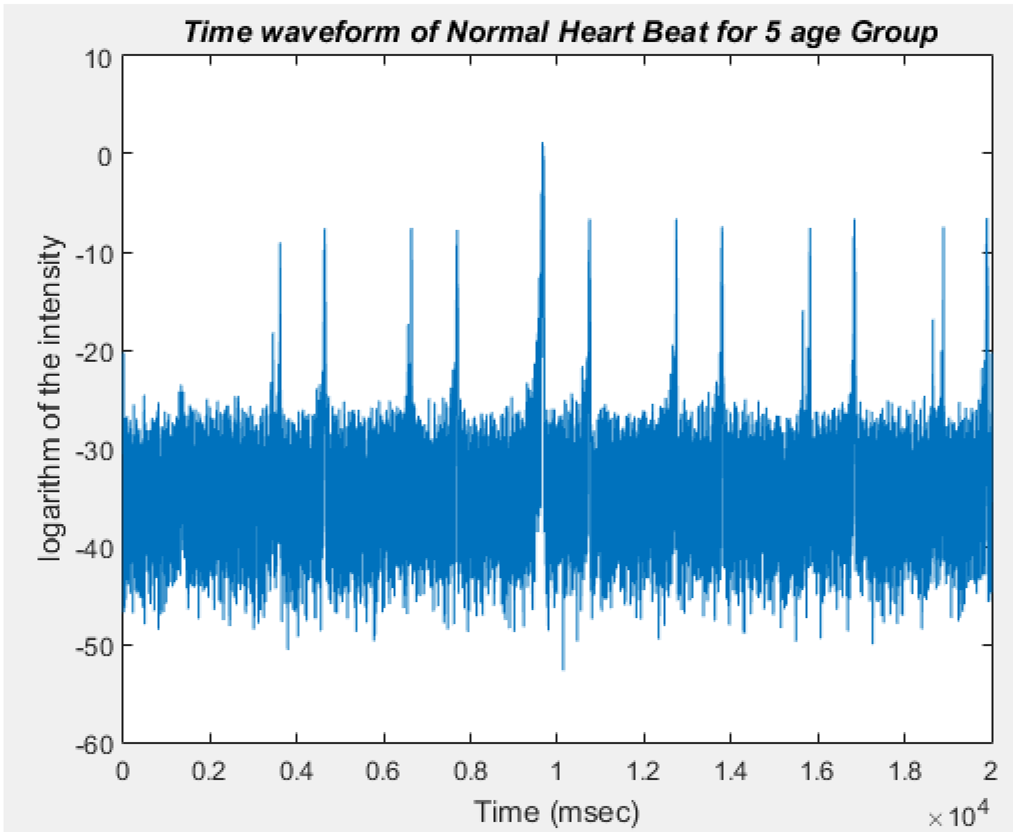

Figure 9: Time waveform for normal heart beat of sample n_2

Figure 9 shows the plot of logarithm of intensity of the sound signal as a function of time. Normally we get and visualize a signal in time domain meaning that the horizontal axis is time and vertical axis is logarithm of intensity for normal heart beat of a 5 years old patient. It have 6 smaller gaps between peak values of the systole and an average of the smaller gaps of systole 0.1 beat per msec. It have 6 higher gaps between peak values for diastole and an average of the higher gaps 0.2 beat per msec. From the review of literature the normal heart beat per minute on average 130bpm .On this thesis the normal heart beat on average beat per minuts 152bpm.The frequency information is hidden because of the fact that the time waveform reveals the sound signal as a function of time.

4.1.2.2. Line spectrum of sample n_2

The MatLab code that transforms the time domain signal to frequency domain, filters the line spectrum of normal heart beat of sample n_2 of human recorded and the child aged 5.

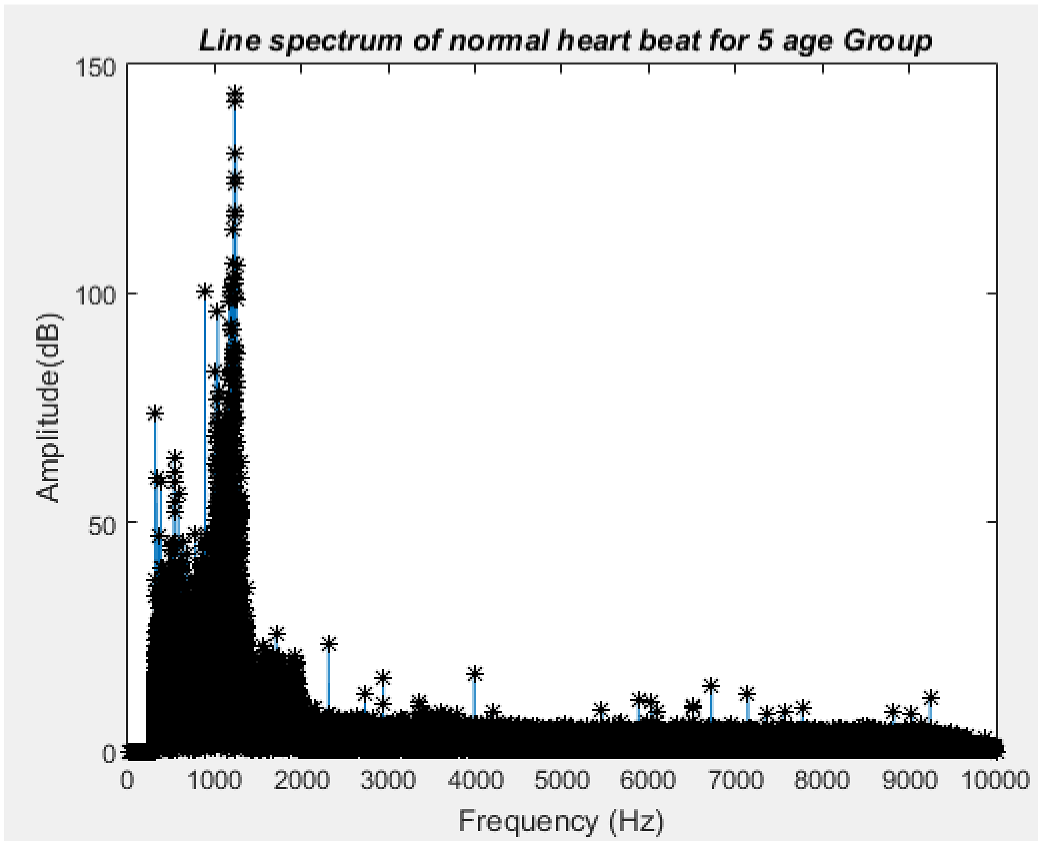

Figure 10: Line spectrum for normal heart beat of sample n_2

Figure 10 shows, the amplitude of the sound signal as a function of frequency. This function was performed after the sound signal is converted from the time domain to the frequency domain by FFT. This plot also shows that the frequency is 1050 Hz and the maximum amplitude approximately 140dB.

4.1.2.3. PSD of sound of sample n_2

Power spectral density (PSD) is a technique of sound analysis and it shows how the density of power of the sound signal is spread over the frequency. It establishes the amount of power on each frequency. This is done after conversion from time to frequency domain and is performed by the MatLab code written in appendix B, filters using low pass filter (LPF) to remove unwanted sound and plots the power verses frequency (PSD) as shown below:

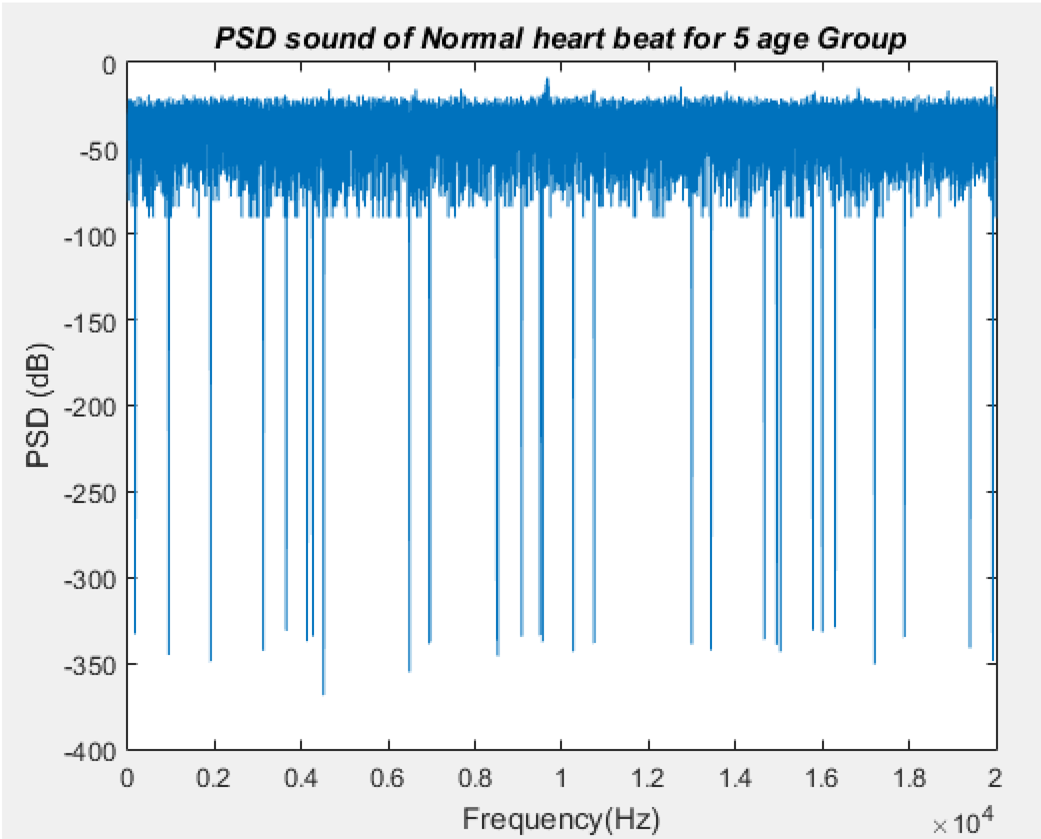

Figure 11 : PSD of sound for normal heart beat of sample n_2

Figure 11 shows, the distribution of the power of the signal over the frequency. One can easily see from the plot the intensity of sound signal has maximum value at approximately -25 dB and 10kHz.

4.1.3. Normal heart beat of sample n_3

4.1.3.1. Time waveform of sample n_3

Even though it conveys little information about the time wave form representation of the normal heart beat of sample n_3 of human recorded for 2 age group. This can be shown by inspecting the wave form of sound produced by the normal heart beat of sample n_3.The MatLab code in appendix C is used for normal heart beat of sample n_3 as shown in Figure 12.

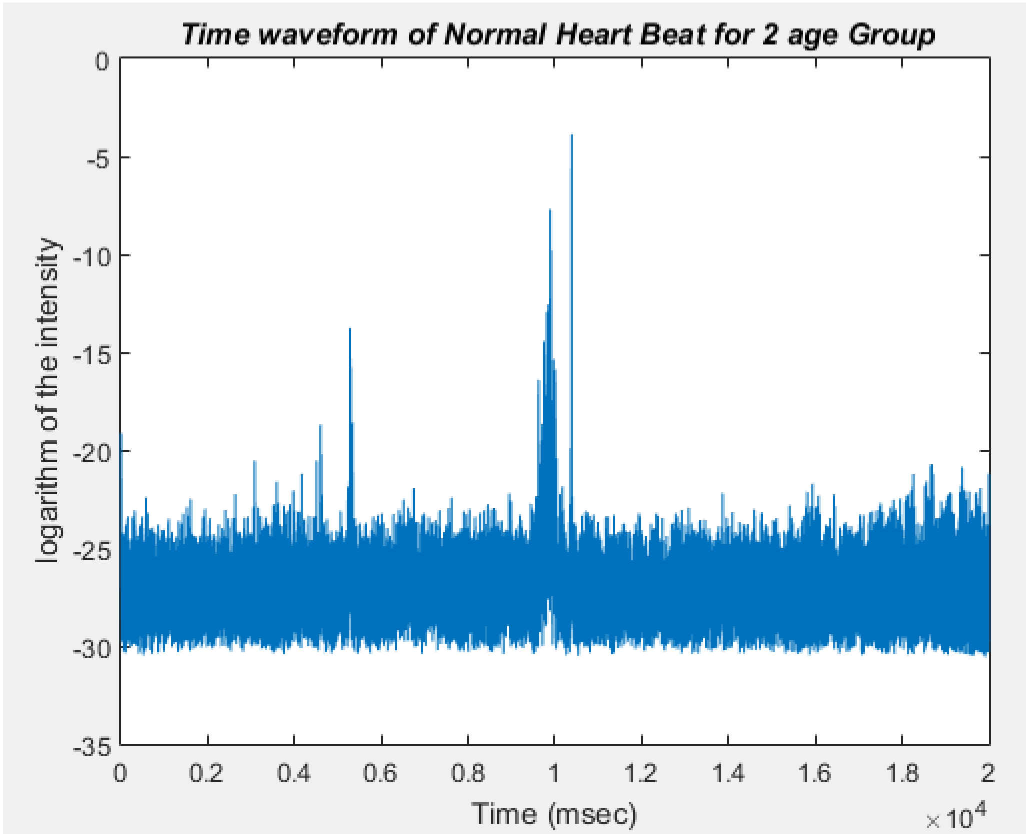

Figure 12 : Time waveform for normal heart beat of sample n_3

Figure 12 shows the plot of sound signal as a function of time. The plot clearly shows the normal heart beat of a child aged 2.There are different peak values correspendes to lub, marks the beginning and end of systole precedes carotid pulse. It have different smaller gaps.The 2nd heart sound dub, marks the end of systole follows carotid pulse. The graph is so overcrowded that it conveys little meaningful information.

4.1.3.2. Line spectrum of sample n_3

The technique that plots of amplitude versus frequency of the sound signal which is converted from time to frequency domain.The MatLab code that transforms in appendix C and plots its line spectrum of normal heart beatof sample n_3 of sound recorded for 2 yearts old patient.

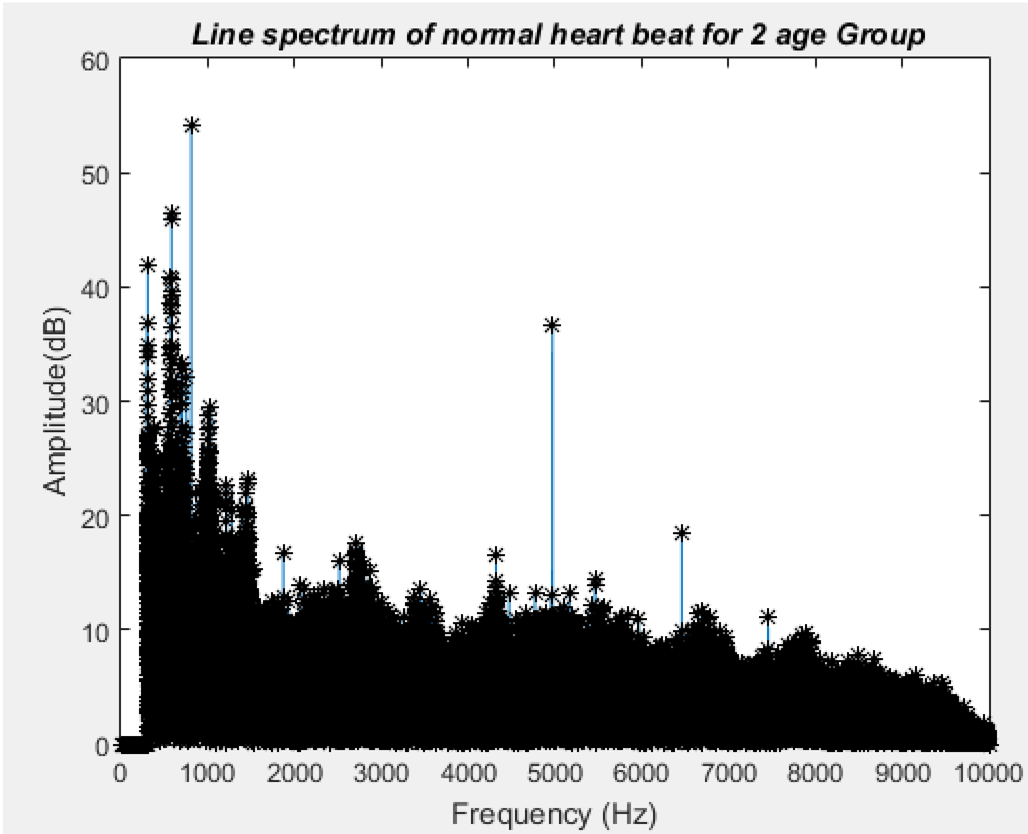

Figure 13 : Line spectrum for normal heart beat of sample n_3

Figure 13 shows the line spectrum of the sound signal of normal heart beat (physiologically split) of human for 2 age group. This plot also shows that the frequency is 950 Hz and the maximum amplitude at 55dB.

4.1.3.3. PSD of sound of sample n_3

This technique of sound analysis is also very essential since it shows how the density of the power of the sound signal is spread over the frequency. The MatLab code written in appendix C which also plots the PSD of this sound signal.

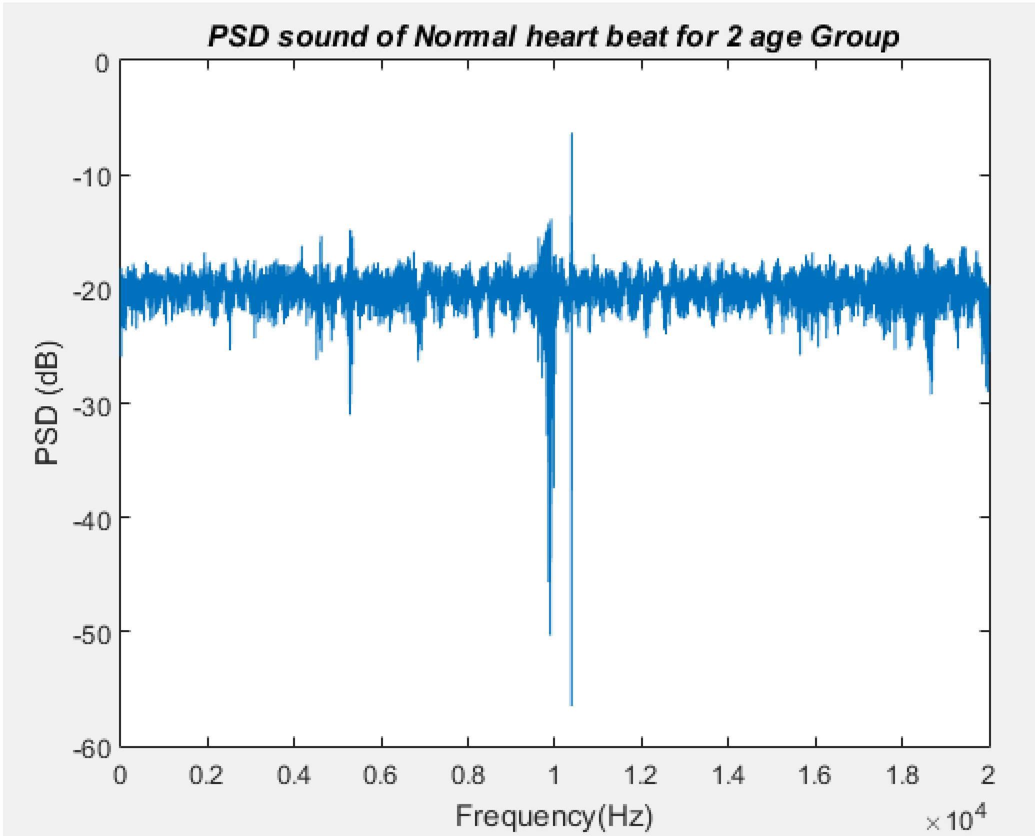

Figure 14 : PSD of sound for normal heart beat of sample n_3

Figure 14 shows the plot of PSD as a function of frequency. The plot clearly shows the distribution of the PSD of sound signal over the frequencies. As observed from the plot, the PSD of the sound signal is -52dB and the dominant frequency value approximately is 10.5KHz.

4.2. Sound of Diastolic Murmur

Three sound of diastolic murmur of human were recorded.

Table 4. Samples taken for diastolic murmur.

Sample	Age
d_1	11
d_2	5
d_3	2

4.2.1. Diastolic murmur of sample d_1

4.2.1.1.Time waveform of sample d_1

The MatLab code appendix D filters the sound signals and plots the time waveform of the sounds of diastolic murmur of sample d_1 of sound recorded for 11 age group as shown Figure 15.

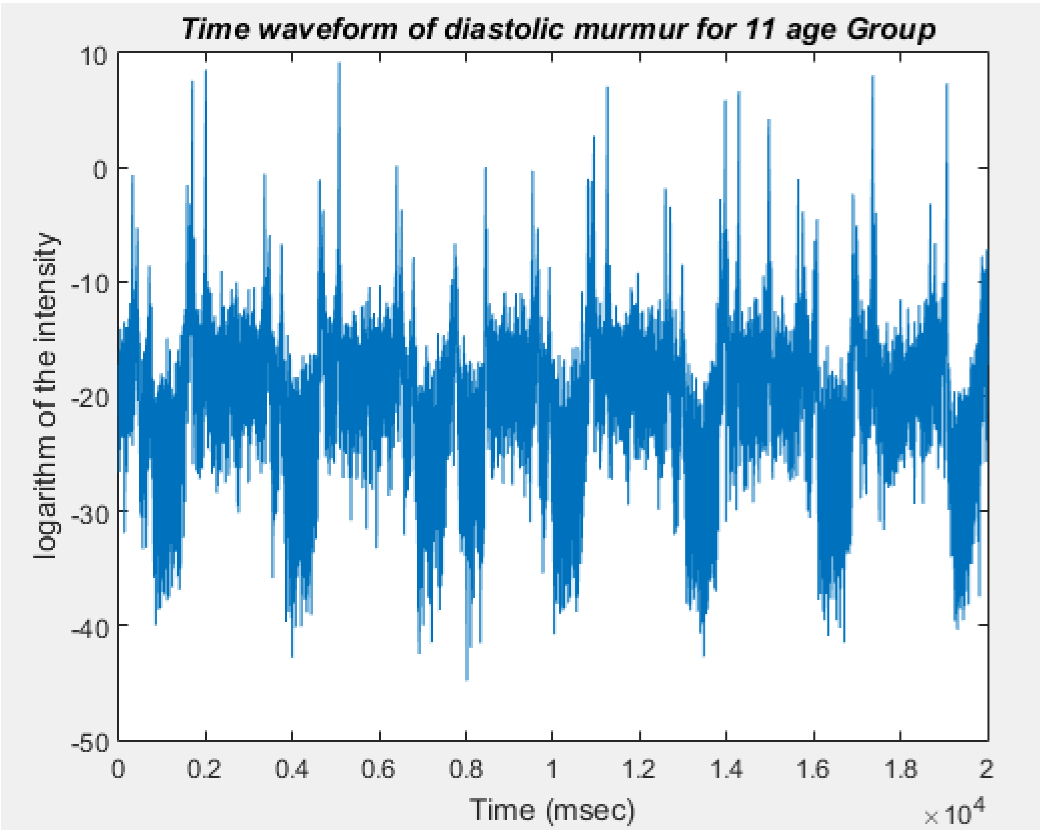

Figure 15 :Time waveform for diastolic murmur of sample d_1

Figure 15 shows, the sound signal of diastolic murmur as a function of time.Visualize the signal in time domain meaning that the horizontal axis is time and vertical axis is logarithm of intensity for diastolic murmur of human for 11 age group. Since the graph is overcrowded, it can only give little meaningful information but most of it is hidden.

4.2.1.2. Line spectrum of sample d_1

The MatLab code appendix D performs the conversion of time dependent signal to frequency dependent signal and plots its line spectrum of diastolic murmur of sample d_1 of sound recorded for 11 age group for visualization.

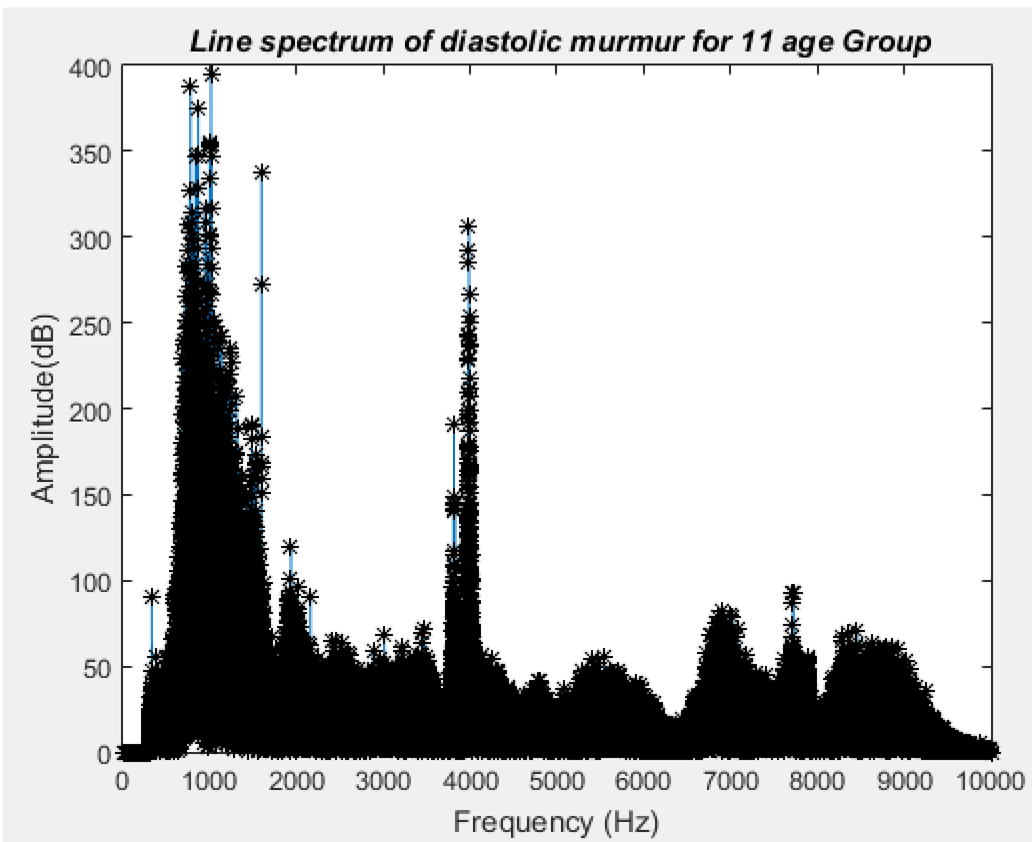

Figure 16 : Line spectrum for diastolic murmur of sample d_1

Figure 16 shows amplitude as a function of frequency. From the plot maximum amplitude approximately at 400Hz frequency with the maximum amplitude 1100dB. The characteristic of mitral stenosis results low pitch, best heard at the cardiac apex and rumbling.

4.2.1.3. PSD of sound of sample d_1

This is the most powerful technique used in sound signal analysis which depicts how the density of the power of the sound signal is spread over the frequency of the signal. It establishes the amount of power of the signal on each signal frequency. This possibility is obtained after the conversion of sound signal from time domain to frequency domain. The MATLAB code which plots the PSD and filters of the sound signal is given in appendix 17.

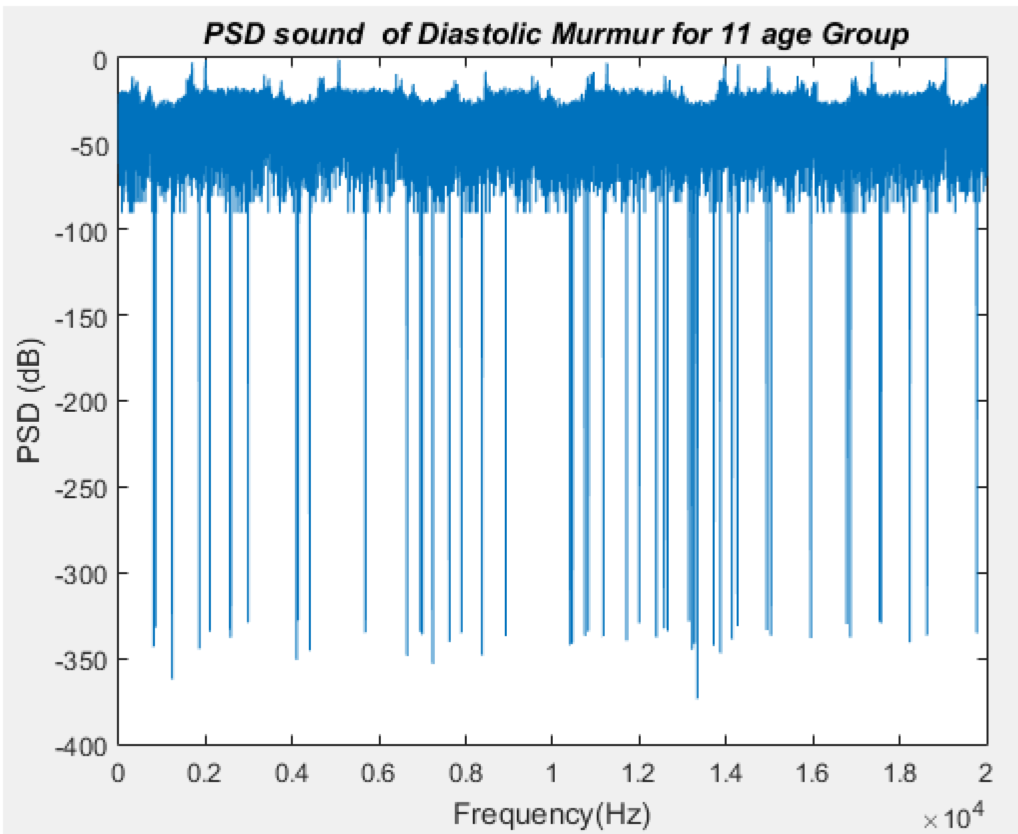

Figure 17 : PSD of sound for diastolic murmur of sample d_1

Figure 17 the above graph depicts the square of the amplitude of the sound signal as a function of frequency. It evidently illustrates the distribution of power of the sound signal over the frequencies. From the plot one can easily observe the density of the power at different frequencies. The PSD of the sound signal is -375 dB and the frequency value is 13.3KHz.

4.2.2. Diastolic murmur of sample d_2

4. 2.2.1. Time waveform of sample d_2

Time waveform representation carries a little information about signal but it is important. The time wave form of diastolic murmur of sample d_2 of sound recorded for 5 age group and plots the time waveform of the sounds of diastolic murmur as shown the Figure 18.

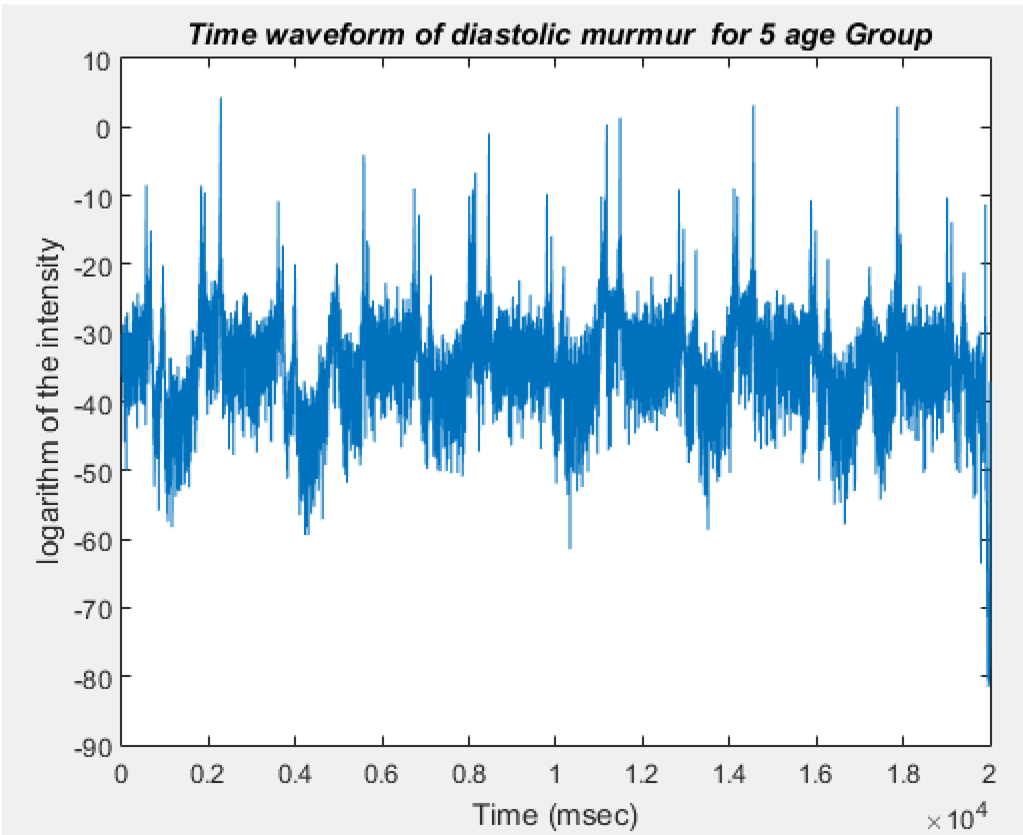

Figure 18 :Time waveform for diastolic murmur of sample d_2

Figure 18 show the sound signal of diastolic murmur as a function of time. The information of diastolic murmur is hidden because of the fact that time waveform reveals the the logarithm of intensity of that signal as a function of time. The required information is not clearly understandable, and conveys poor information.

4.2.2.2. Line spectrum of sample d_2

The conversion of time dependent signal to frequency dependent signal.The MatLab code appendix E that transform of line spectrum of diastolic murmur of sample d_2 for 5 age group.

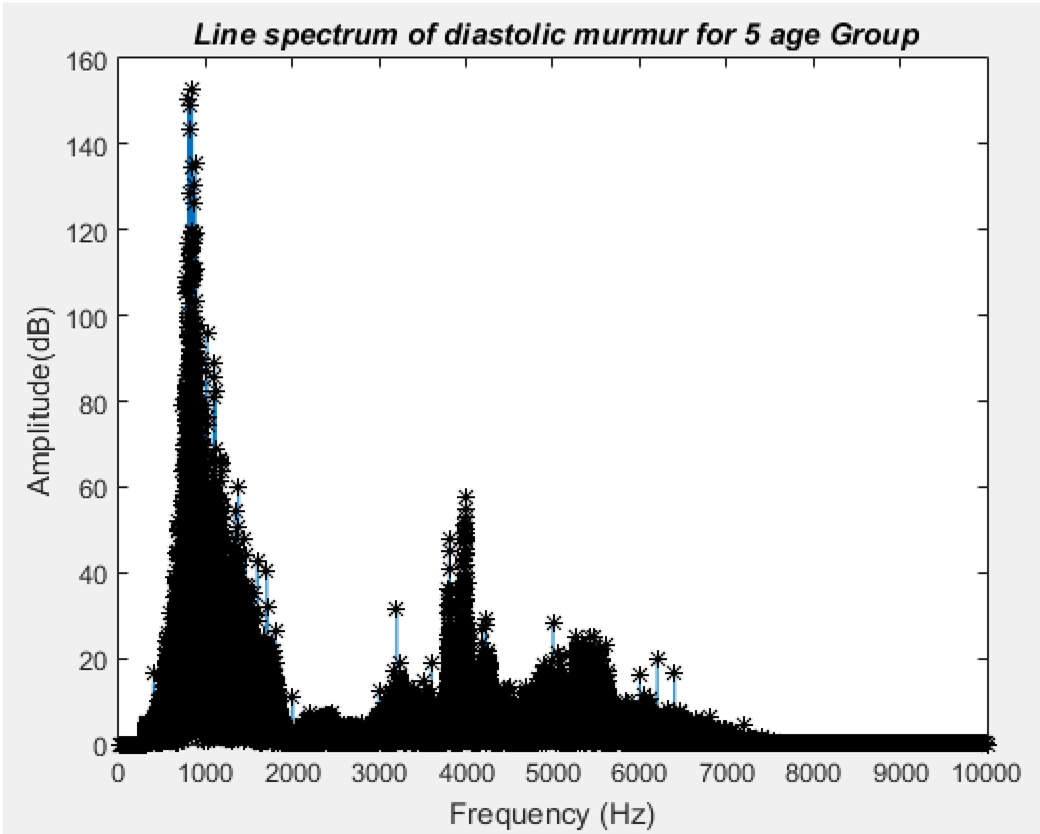

Figure 19: Line spectrum for diastolic murmur of sample d_2

Figure 19 shows amplitude as a function of frequency. The maximum amplitude approximately at 152 Hz frequency with the maximum amplitude exhibited a level which lies 1000dB. There different small peaks at 4000Hz frequency and approximately 60dB maximum amplitude.

4.2.2.3. PSD of sound of sample d_2

The MatLab code given in appendix E transforms the signal in time domain in to frequency domain and plots PSD as shown in Figure 20.

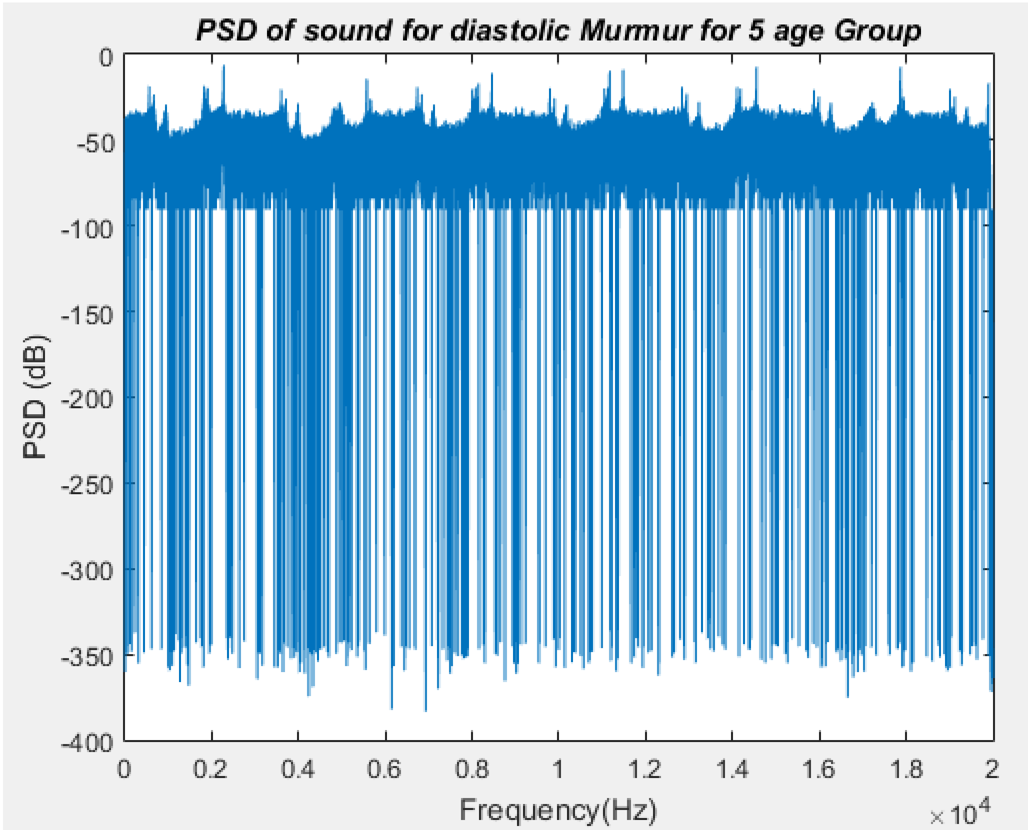

Figure 20 : PSD of sound for diastolic murmur of sample d_2

Figure 20, the plot clearly shows the distribution of the power of the signal over the frequencies. One can easily evaluate the density of the power at different frequencies. This plot also shows that the intensity of the sound signal has maximum value at approximately 7KHz frequency and the PSD of the sound signal is -380 dB.

4.2.3. Diastolic murmur of sample d_3

4. 2.3.1. Time waveform of sample d_3

The sound signals and plots the time waveform of the sounds of diastolic murmur of sample d_3 of sound captured for 2 age group and the MatLab code appendix F used to read the recorded sound for diastolic murmur of sample d_3.

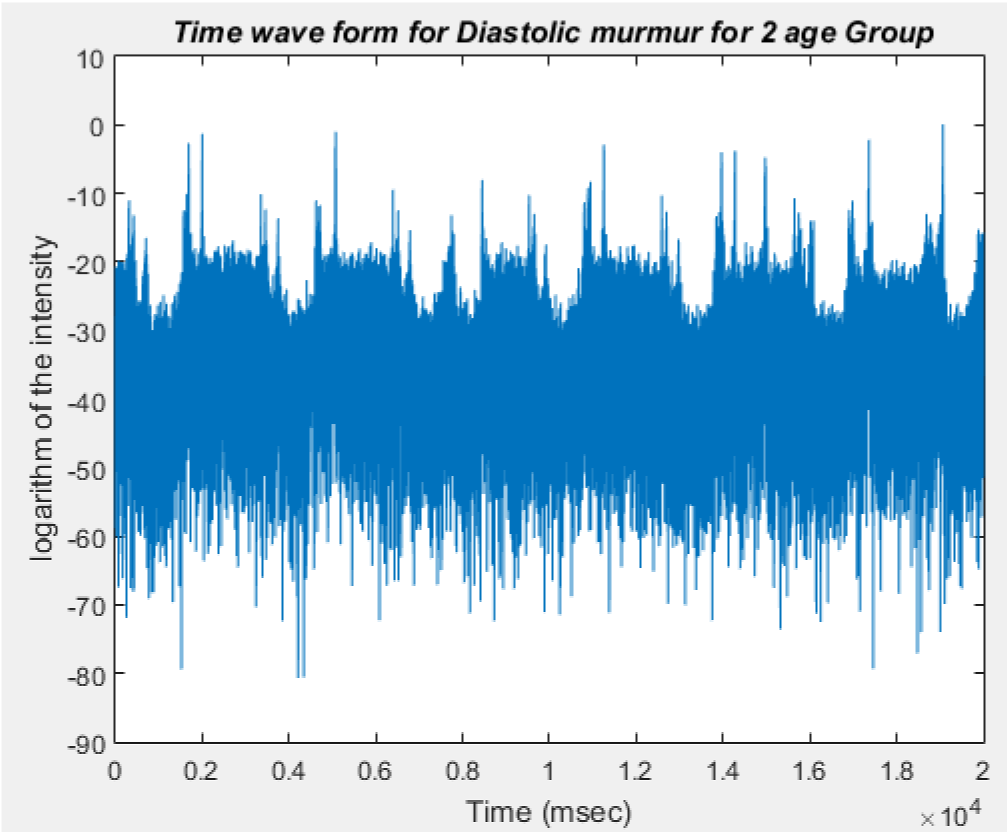

Figure 21:Time waveform for diastolic murmur of sample d_3

Figure 21 shows the sound signal of diastolic murmur as a function of time. The mitral stenosis sounds were the first heart sound S_1(lub) and the second heart sound S_2(dub). The noise sound or murmur was made between the second heart sound S_2(dub) and first heart sound S_1 (lub).

4.3.2.2. Line spectrum of sample d_3

First, using fast Fourier transform (FFT), signal transformation from amplitude versus time domain versus frequency domain is carried out. The MatLab code in appendix F is used to read the recorded sound for diastolic murmur of sample d_3 for 2 age group.

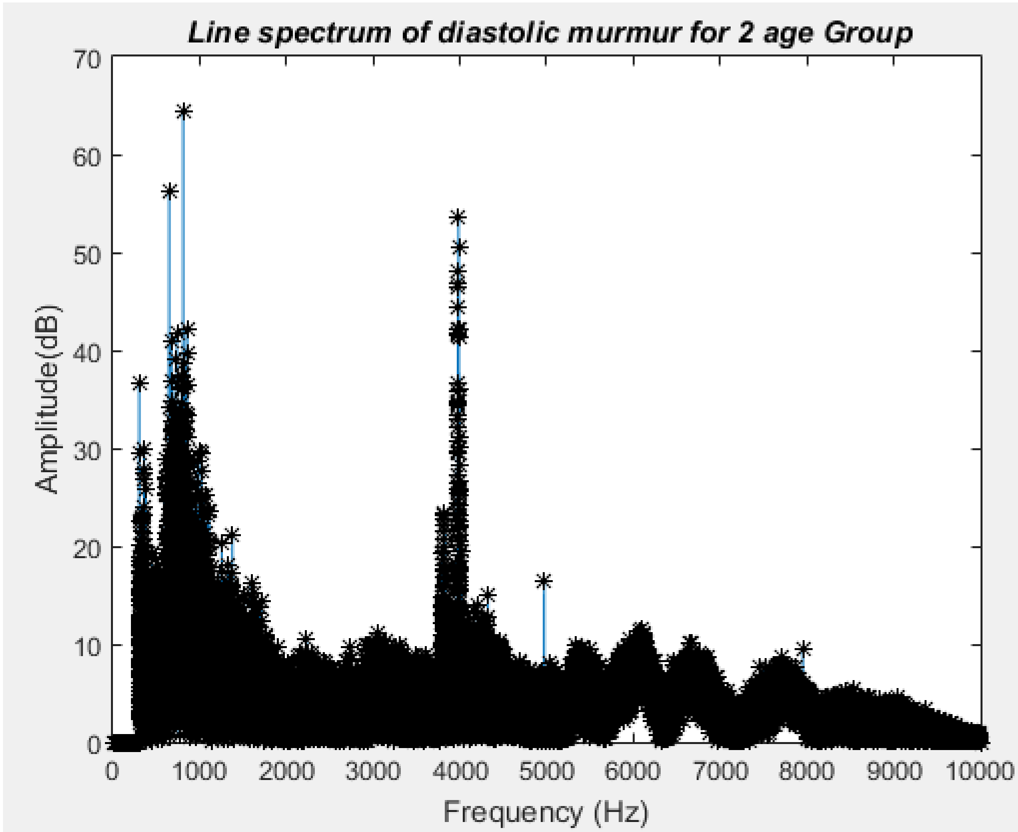

Figure 22 : Line spectrum for diastolic murmur of sample d_3

Figure 22 shows the amplitude as a function of frequency. The plots depicted that the maximum amplitude approximately at 920 Hz frequency with the maximum amplitude is 65 dB. There additional peak values appear at 4000Hz frequency and 55dB amplitude of the sound signal.

4.2.3.3. PSD of sound of sample d_3

The MatLab code given in appendix F transforms the signal in time domain in to frequency domain and plots PSD as shown in Figure 23.

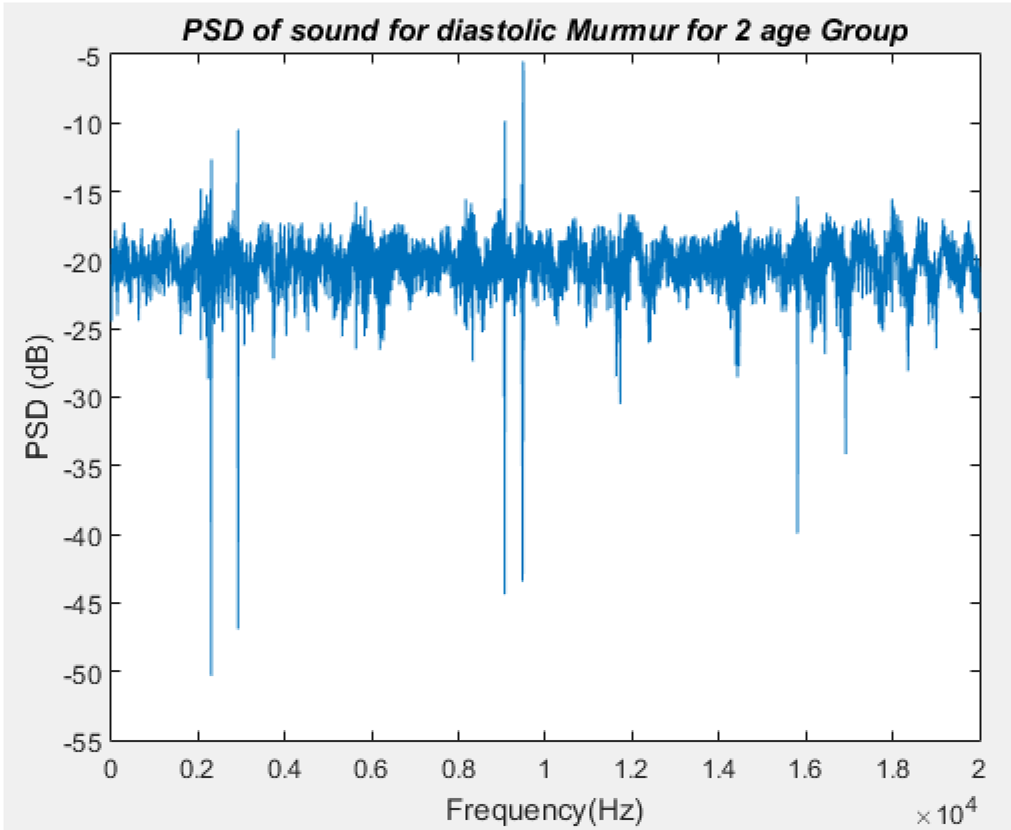

Figure 23 : PSD of sound for diastolic murmur of sample d_3

Figure 23 shows the PSD of sound signal as a function of frequency. As we can see from the plot, the PSD of the sound signal is -50dB which is acquired at a dominant frequency value approximately is 2.2KHz.

4.3. Sound of Systolic Murmur

Three sounds of systolic murmur of human were recorded.

Table 5. Samples taken for systolic murmur

Sample	Age
t_1	11
t_2	5
t_3	2

4.3.1. Systolic murmur of sample t_1

4.3.1.1. Time waveform of sample t_1

The time waveform below being plotted by MatLab code Appendix G shows you the sound due to systolic murmur of sample t_1 of sound recorded 11years old patient.

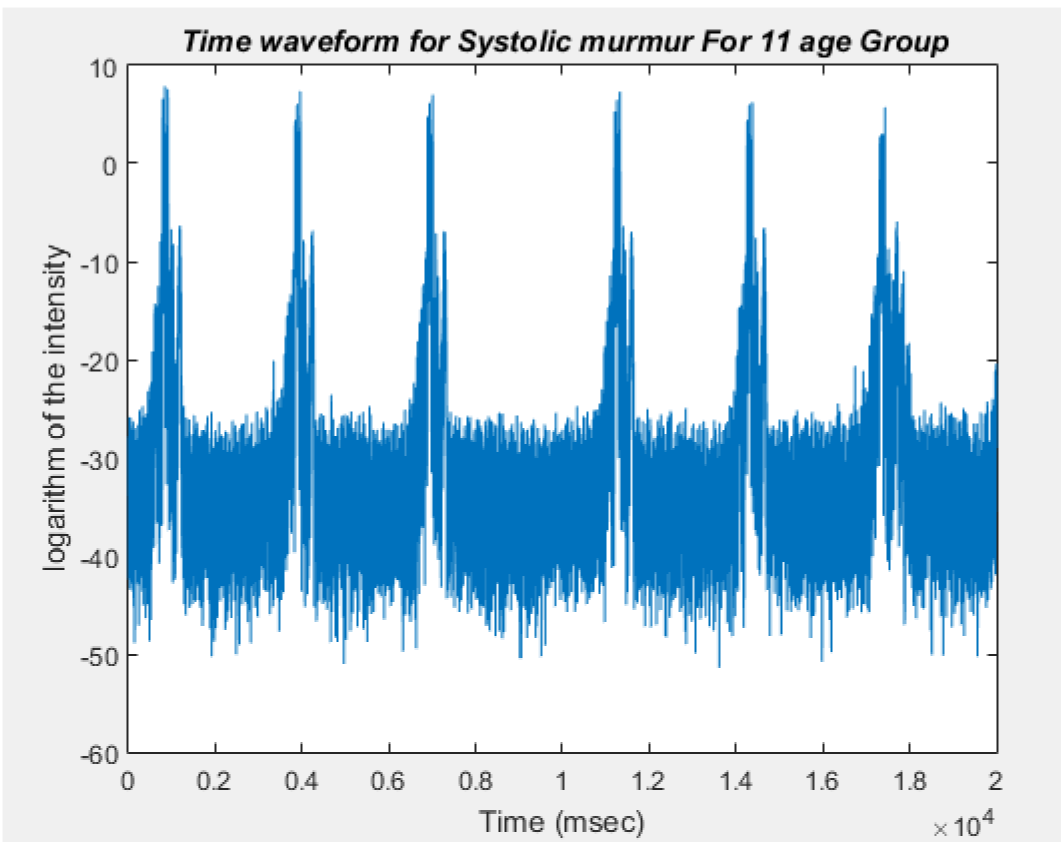

Figure 24 :Time waveform for systolic murmur of sample t_1

Figure 24 shows the plot logarithm of intensity of time waveform for systolic muemur of sample t_1 of human as a function of time. Normally we get and visualize a signal in time domain meaning that the horizontal axis is time and vertical axis is logarithm of intensity for systolic murmur (mitral regurgitation) of human for 11 age group. The mitral regurgitation or pan systolic murmur thought out systolic period causes high pitch whistling murmur due to high velocity.This murmur sound like BuRR, BuRR had been presented between the first heart sound S_1(lub) and the second heart sound S_2(dub) .

4.3.1.2. Line spectrum of sample t_1

The sound signal of systolic murmur of sample t_1 of sound recorded of a child aged 11. It is very important to clearly show the hidden information contained in the sound signal. This information is observed by applying the MatLab code Appendix G which plots its spectrum.

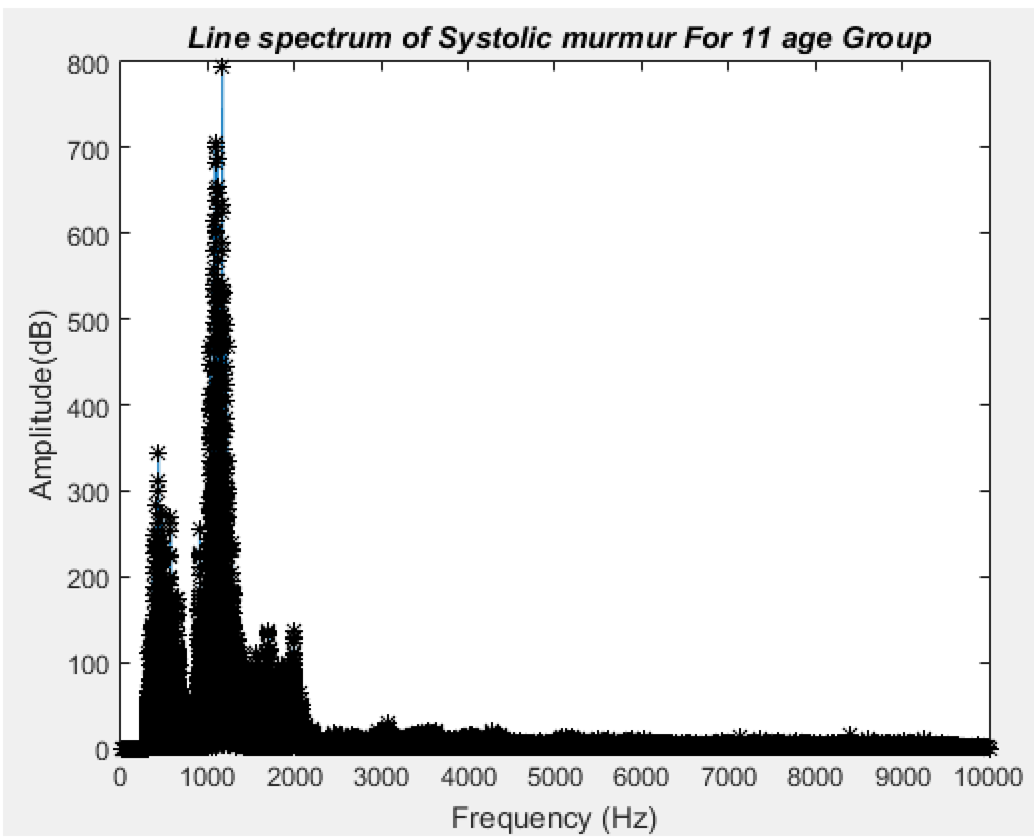

Figure 25 : Line spectrum for systolic murmur of sample t_1

Figure 25 shows the amplitude of sound signal as a function of frequency. From the plot the maximum amplitude approximately is 1200Hz frequency with the maximum amplitude at

800dB. The systolic murmur (mitral regurgitation) was described as a high-pitch, blowing holosystolic murmur best heard at apex.

4.3.1.2. PSD of sound of sample t_1

This technique of sound signal display is preferable to illustrate the power distribution over different frequencies. The plot that shows the power spectral density of a signal is plotted below using MATLAB code Appendix G shows the systolic murmur of sample t_1 of sound for 11 age group.

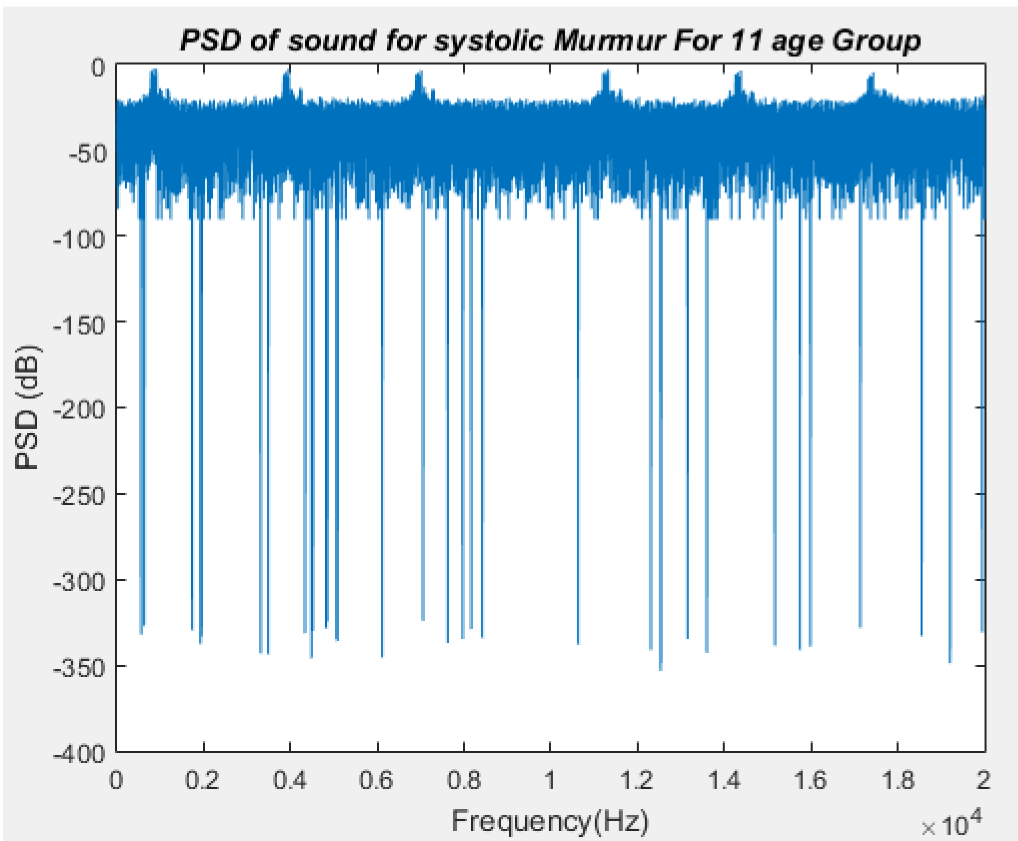

Figure 26 :PSD of sound for systolic murmur of sample t_1

Figure 26 shows the PSD of sound signal as a function of frequency. As observed from the plot, the PSD of the sound signal is -350 dB and the frequency range value is 12.5KHz.

4.3.2. Systolic murmur of sample t_2

4.3.2.1. Time waveform of sample t_2

The time waveform of systolic murmur of sample t_2 recorded of a child aged 5 and being plotted by MatLab code Appendix H shows you the sound .

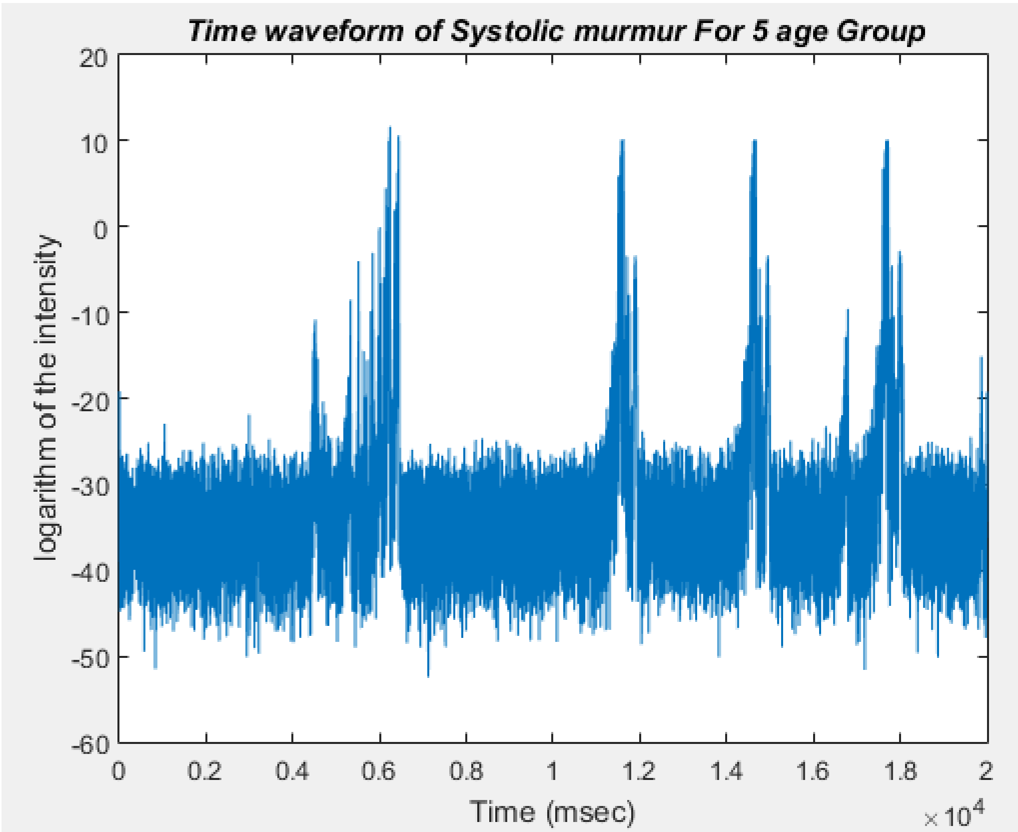

Figure 27:Time waveform for systolic murmur of sample t_2

Figure 27 shows the sound signal as a function of time. The mitral regurgitation caused by the turbulent flow through the incompetent mitral valve leafles in to the left atrium. The mitral regurgitation or pan systolic murmur thought out systolic period causes high pitch whistling murmur due to high velocity.

4.3.2.2. Line spectrum of the sample t_2

Line spectrum visualization technique of sound signal provides detail information of the signal than the time wave form. To make known this information, first the signal must be converted from time domain to frequency domain using FFT. The conversion is obtained by the MatLab code given in appendix H.

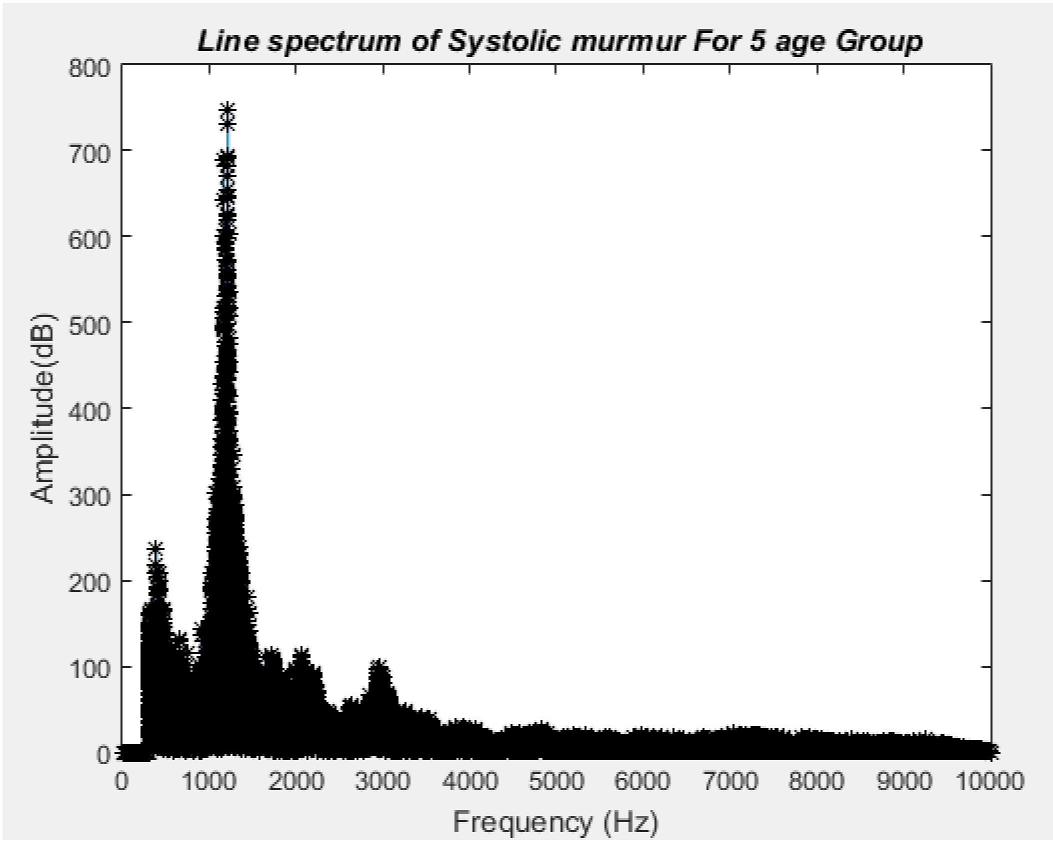

Figure 28 : Line spectrum for systolic murmur of sample t_2

Figure 28 the amplitude of sound signal as a function of frequency. This plot shows that the maximum amplitude approximately is 1250Hz frequency with the maximum amplitude at 750dB. There is additional peak of systolic murmr of human with the frequency of 500Hz and 250dB maximum amplitude.

4.3.2.3. PSD of sound of sample t_2

The PSD of sound below being plotted by MatLab code Appendix H shows you the sound due to systolic murmur of sample t_2 and sound recorded a child aged 5 .

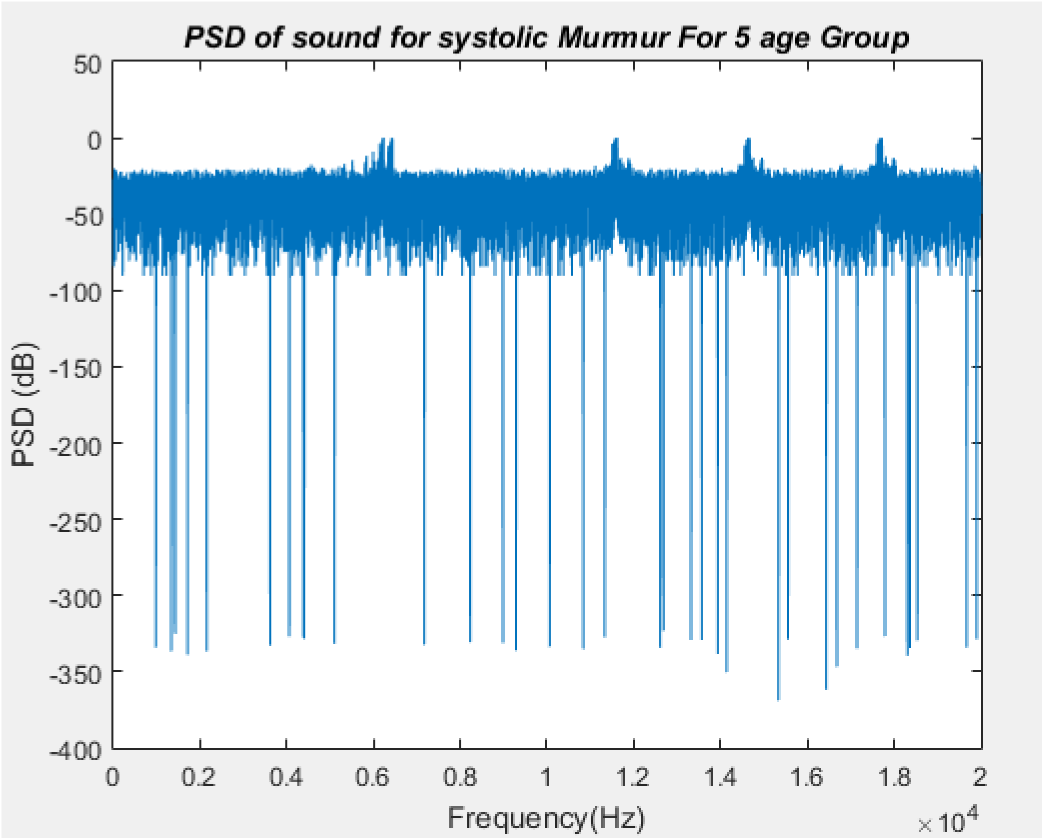

Figure 29 :PSD of sound for systolic murmur of sample t_2

Figure 29 shows the PSD of sound signal as a function of frequency. As observed from the plot, the PSD of the sound signal is -370 dB which is acquired at a dominant frequency value approximately is 15.2KHz.

4.3.3. Systolic murmur of sample t_3

4.3.3.1. Time waveform of sample t_3

The time waveform below being plotted by MatLab code Appendix I shows the sound due to systolic murmur of sample t_3 of sound captured and the child aged 2.

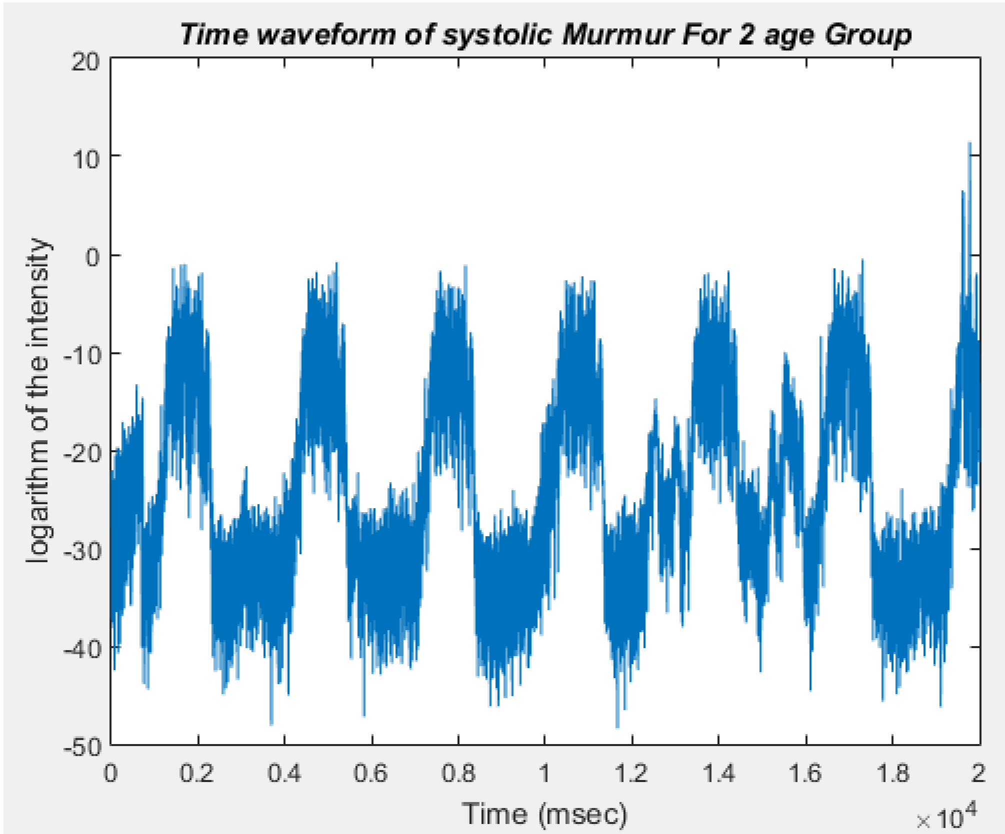

Figure 30 :Time waveform for systolic murmur of sample t_3

Figure 30 shows the sound signal as a function of time. The mitral regurgitation caused by the turbulent flow through the incompetent mitral valve leafles in to the left atrium. The mitral regurgitation or pan systolic murmur thought out systolic period causes high pitch whistling murmur due to high velocity.

4.3.3.2. Line spectrum of sample t_3

Line spectrum representation is more reasonable than time wave form. The hidden information in time wave form can be easily visualized in line spectrum plot. The MatLab code Appendix I which plots its spectrum .

.

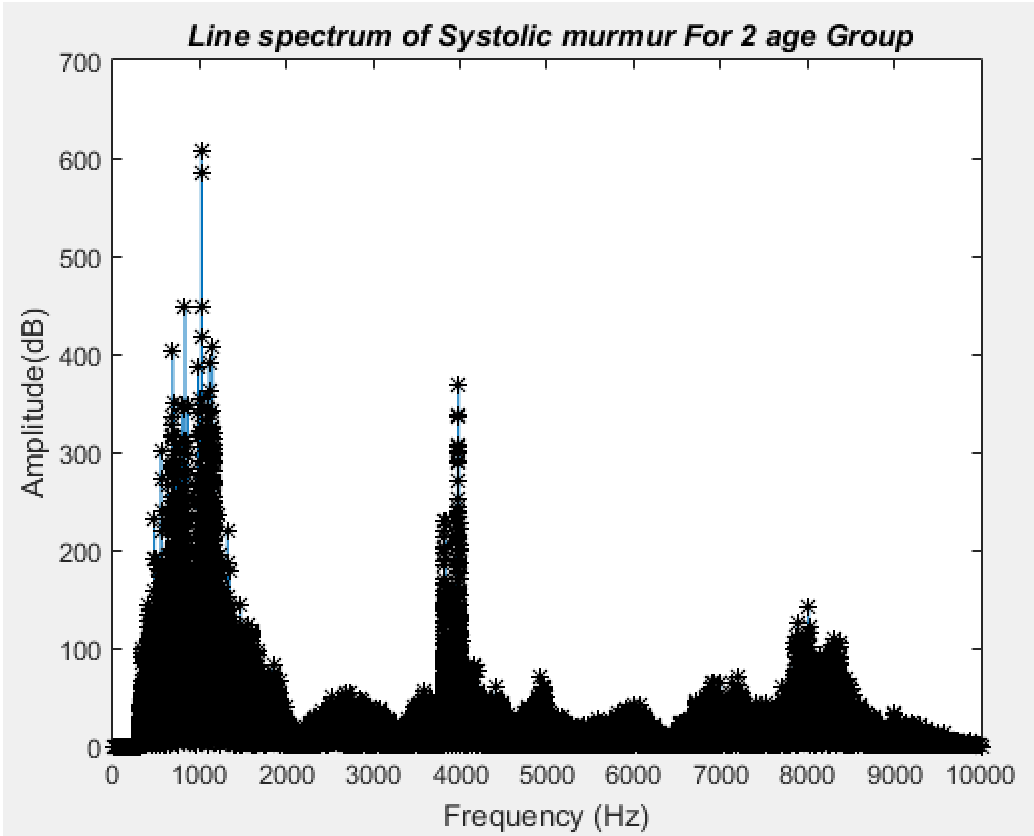

Figure 31 : Line spectrum for systolic murmur of sample t_3

Figure 31 shows that, the amplitude of sound signal as a function of frequency. As observed from the plots, the frequency is 1100Hz and the maximum amplitude at 615dB. There were different additiona peak values of maximum amplitude at different frequency. So at 4000Hz,8000Hz frequency range the maximum amplitude of systolic murmur of sample t_3 400dB and 150dB respectively. The systolic murmur (mitral regurgitation) was described as a high-pitch, blowing holosystolic murmur best heard at apex.

4.3.3.3. PSD of sound of sample t_3

Power spectral density is frequently used to illustrate the density of power at different frequency. MATLAB code transforms the sound signal from the time function into the frequency function through the use of FFT and plots its power spectral density.

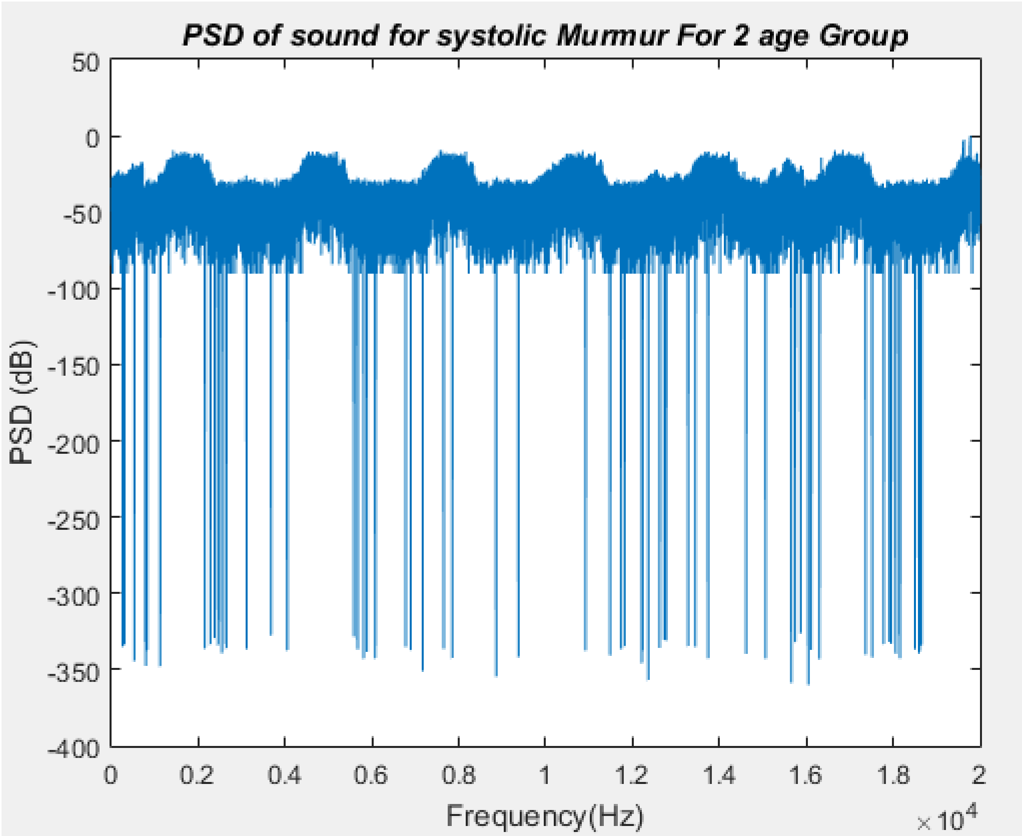

Figure 32 :PSD of sound for systolic murmur of sample t_3

Figure 32 shows the PSD of sound signal as a function of frequency. From the plot one can easily observe the density of the power at different frequencies. As observed from the plot, the PSD of the sound signal is -360dB which is acquired at a dominant frequency value approximately is 16KHz.

5. SUMMARY, CONCLUSIONS AND COMMENDATIONS

5.1. SUMMARY

Three sounds were recorded for this study. They were three sounds from normal heart beat which is named as “sample n_1”,“sample n_2”,“sample n_3”, three sounds for diastolic murmur which is“ sample d_1”,“sample d_2”,“sample d_3”and three sounds for systolic murmur which is “sample t_1”,“sample t_2”,“sample t_3”. Those sounds were recorded at 11,5 and 2 age groups respectively. Comparisons have been made between the sounds of normal heart beat (physiologically split), sound due to diastolic murmur (mitral stenosis) and systolic murmur (mitral regurgitation) from the 11age group,5 age groupe and 2age groupe are clear difference detected or identified. This work utilized three different plots time waveform, line spectrum and PSD. Line spectrum and PSD are in frequency domain but time waveform is in time domain. All the sound data were captured at equal sampling frequency and have equal number of samples. All of them are used to distinguish the abnormal heart sounds of 0-15 age group level of heart sound.

5.2. CONCLUSIONS

For this study, three types of sound sources and each sound signal were three samples.These sond were directly recorded in to Matlab software using the function wavrecord (Ns, fs, NC,'Td'). This work made use of three different plots named: Time waveform, Line spectrum and Power Spectral Density (PSD).The time waveform technique for each of the three sound signals was plotted as logarithm of intensity versus time. As seen from the graphs, was not enough information about the signals. Time wave form was plotted in the time domain.

Line spectrum is another sound signals plotting technique used the plotten spectrum of one can determine the amplitudes of the signals at each frequency. The plot was done after the transformation of the signals from time domain to frequency domain using FFT. The plot is the amplitude of the signal versus frequency. It clearly details the information contained in the sound signals. This enables us to identify that the sound signals which have different maximum amplitudes at different frequencies.

PSD is also one of the plots that this thesis work was used. The plot depicts the intensity level of the sound signal versus frequency. The plot was done after the conversion of time dependent to frequency dependent, which is facilitated by FFT. In other words, it illustrated the intensity of the sound signals at different frequencies. Therefore, established the relationship between the sounds produced due to abnormal heart sound (distolic and systolic murmur) with the same age group.Each sound signal has frequency value for its maximum amplitude. One can conclude that the sound signal produced by a particular heart beat gives a sound which specifically has certain spectrum.

5.3. RECOMMENDATIONS

The study showed that sound signal obtained from the normal heart beat of human due to the abnormal heart sound (diastolic and systolic) murmur were found to have different spectral representation. Each of the sound signals has frequency value for their maximum of the amplitude. This is used to conclude that the sound signal produced by a particular heart sound yields which is specifically has certain spectrum. Therefore, the researcher recommends that it is important to capture the sound signal produced due to the abnormal of heart sound and plot their spectra.

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7. APPENDICES

Appendix A: MATLAB code for normal heart beat(4)

```
close all
clear all
%fid = fopen('cday.wav');
L = audioread('normal4.wav');
nn = length(L);             %Lengthofsignal
Fs = 20000;                 % Sampling frequency
T = 1/Fs;                   % Sample time
t = (0:nn-1)*T;             % Time vector
NFFT = 2^nextpow2(nn); % Next power of 2 from length of y
Y = fft(L,NFFT);
f = Fs/2*linspace(0,1,NFFT/2+1);
SegmentLength = NFFT;
[P,F] = pwelch(Y,ones(NFFT,0),1,NFFT,Fs,'power');
figure,
Y(f<=300|f>=10000) = 0;
plot(F,10*log10(P))
title('\bf\it Time waveform of Normal Heart Beat for 11 age groupe ')
xlabel('Time (msec)')
ylabel('logarithm of the intensity')
figure
plot(f,2*abs(Y(1:NFFT/2+1)));
hold on
plot(f,2*abs(Y(1:NFFT/2+1)),'k*');
title('\bf\it Line spectrum  of normal heart beat for 11 age groupe ')
xlabel('Frequency (Hz)')
ylabel('Amplitude(dB)')
% Power spectrum is computed when you pass a 'power' flag input
[P,F] = pwelch(Y,ones(SegmentLength,1),0,NFFT,Fs,'power');
figure,
plot(F,10*log10(P))
title('\bf\it PSD sound  of  Normal heart beat for 11 age groupe ')
xlabel('Frequency(dB)')
ylabel('PSD (dB)')
```

Appendix B: MATLAB code for normal heart beat(5)

```
close all
clear all
%fid = fopen('cday.wav');
L = audioread('normal5.wav');
nn = length(L);             % Length of signal
Fs = 20000;                 % Sampling frequency
```

```
T = 1/Fs;                    % Sample time
t = (0:nn-1)*T;              % Time vector
NFFT = 2^nextpow2(nn); % Next power of 2 from length of y
Y = fft(L,NFFT);
f = Fs/2*linspace(0,1,NFFT/2+1);
SegmentLength = NFFT;
[P,F] = pwelch(Y,ones(NFFT,0),1,NFFT,Fs,'power');
figure,
Y(f<=300|f>=10000) = 0;
plot(F,10*log10(P))
title('\bf\it Time waveform of  Normal Heart Beat for 5 age groupe ')
xlabel('Time (msec)')
ylabel('logarithm of the intensity')
figure
plot(f,2*abs(Y(1:NFFT/2+1)));
hold on
plot(f,2*abs(Y(1:NFFT/2+1)),'k*');
title('\bf\it Line spectrum  of normal heart beat for 5 age groupe ')
xlabel('Frequency (Hz)')
ylabel('Amplitude(dB)')
% Power spectrum is computed when you pass a 'power' flag input
[P,F] = pwelch(Y,ones(SegmentLength,1),0,NFFT,Fs,'power');
figure,
plot(F,10*log10(P))
title('\bf\it PSD of sound of Normal heart beat for 5 age groupe ')
xlabel('Frequency(dB)')
ylabel('PSD (dB)')
```

Appendix C: MATLAB code for normal heart beat(6)

```
close all
clear all
%fid = fopen('cday.wav');
L = audioread('normal6.wav');
nn = length(L);              % Length of signal
Fs = 20000;                  % Sampling frequency
T = 1/Fs;                    % Sample time
t = (0:nn-1)*T;              % Time vector
NFFT = 2^nextpow2(nn); % Next power of 2 from length of y
Y = fft(L,NFFT);
f = Fs/2*linspace(0,1,NFFT/2+1);
SegmentLength = NFFT;
[P,F] = pwelch(Y,ones(NFFT,0),1,NFFT,Fs,'power');
figure,
Y(f<=300|f>=10000) = 0;
plot(F,10*log10(P))
title('\bf\it Time waveform for Normal Heart Beat for 2 age groupe ')
```

```
xlabel('Time (msec)')
ylabel('logarithm of the intensity')
figure
plot(f,2*abs(Y(1:NFFT/2+1)));
hold on
plot(f,2*abs(Y(1:NFFT/2+1)),'k*');
title('\bf\it Line spectrum  of  normal heart beat for 2 age groupe ')
xlabel('Frequency (Hz)')
ylabel('Amplitude(dB)')
% Power spectrum is computed when you pass a 'power' flag input
[P,F] = pwelch(Y,ones(SegmentLength,1),0,NFFT,Fs,'power');
figure,
plot(F,10*log10(P))
title('\bf\it PSD sound of Normal heart beat for 2 age groupe ')
xlabel('Frequency(dB)')
ylabel('PSD (dB)')
```

Appendix D: MATLAB code for plotting For Diastolic murmur(8)

```
close all
clear all
%fid = fopen('cday.wav');
L = audioread('diastolic8.wav');
nn = length(L);                % Length of signal
Fs = 20000;                    % Sampling frequency
T = 1/Fs;                      % Sample
t = (0:nn-1)*T;                % Time vector
NFFT = 2^nextpow2(nn); % Next power of 2 from length of y
Y = fft(L,NFFT);
f = Fs/2*linspace(0,1,NFFT/2+1);
SegmentLength = NFFT;
[P,F] = pwelch(Y,ones(SegmentLength,0),1,NFFT,Fs,'power');
figure,
Y(f<=300|f>=10000) = 0;
plot(F,10*log10(P))
title('\bf\it Time waveform of  Diastolic Murmur for 11 age groupe ')
xlabel('Time (msec)')
ylabel('logarithm of the intensity')

figure
plot(f,2*abs(Y(1:NFFT/2+1)));
hold on
plot(f,2*abs(Y(1:NFFT/2+1)),'k*');
title('\bf\it Line spectrum  of  Diastolic Murmur for 11 age groupe ')
```

```
xlabel('Frequency (Hz)')
ylabel('Amplitude(dB)')
figure,
plot(F,10*log10(P))
title('\bf\itPSD of sound for Diastolic Murmur for 11 age groupe ')
xlabel('frequency(Hz)')
ylabel('PSD(dB)')
```

Appendix E: MATLAB code for plotting For Diastolic murmur(09)

```
close all
clear all
%fid = fopen('cday.wav');
L = audioread('diastolic09.wav');
nn = length(L);  % Length of signal
Fs = 20000;                    % Sampling frequency
T = 1/Fs;                     % Sample time
t = (0:nn-1)*T;                % Time vector
NFFT = 2^nextpow2(nn); % Next power of 2 from length of y
Y = fft(L,NFFT);
f = Fs/2*linspace(0,1,NFFT/2+1);
SegmentLength = NFFT;
[P,F] = pwelch(Y,ones(SegmentLength,0),1,NFFT,Fs,'power');
figure,
Y(f<=300|f>=1000) = 0;
plot(F,10*log10(P))
title('\bf\it Time waveform for Diastolic Murmur for 5 age groupe ')
xlabel('Time (msec)')
ylabel('logarithm of the intensity')

figure
plot(f,2*abs(Y(1:NFFT/2+1)));
hold on
plot(f,2*abs(Y(1:NFFT/2+1)),'k*');
title('\bf\it Line spectrum  for Diastolic Murmur for 5 age groupe ')
xlabel('Frequency (Hz)')
ylabel('Amplitude(dB)')
figure,
plot(F,10*log10(P))
title('\bf\itPSD of sound for Diastolic Murmur for  age groupe ')
xlabel('frequency(Hz)')
ylabel('PSD(dB)')
```

Appendix F: MATLAB code for plotting For Diastolic murmur(15)

```
close all
clear all
%fid = fopen('cday.wav');
L = audioread('diastolic15.wav');
nn = length(L);             % Length of signal
Fs = 20000;                 % Sampling frequency
T = 1/Fs;                   % Sample time
t = (0:nn-1)*T;             % Time vector
NFFT = 2^nextpow2(nn); % Next power of 2 from length of y
Y = fft(L,NFFT);
f = Fs/2*linspace(0,1,NFFT/2+1);
SegmentLength = NFFT;
[P,F] = pwelch(Y,ones(SegmentLength,0),1,NFFT,Fs,'power');
figure,
Y(f<=300|f>=10000) = 0;

plot(F,10*log10(P))
title('\bf\it Time waveform of  Diastolic Murmur for 2 age groupe ')
xlabel('Time (msec)')
ylabel('logarithm of the intensity')

figure
plot(f,2*abs(Y(1:NFFT/2+1)));
hold on
plot(f,2*abs(Y(1:NFFT/2+1)),'k*');
title('\bf\it Line spectrum  for Diastolic Murmur for 2 age groupe ')
xlabel('Frequency (Hz)')
ylabel('Amplitude(dB)')
figure,
plot(F,10*log10(P))
title('\bf\itPSD of sound for Diastolic Murmur for 2 age groupe ')
xlabel('frequency(Hz)')
ylabel('PSD(dB)')
```

Appendix G: MATLAB code for plotting For Systolic murmur (29)

```
close all
clear all
%fid = fopen('cday.wav');
L = audioread('systolic29.wav');
nn = length(L);             % Length of signal
```

```
Fs = 20000;                  % Sampling frequency
T = 1/Fs;                   % Sample time
t = (0:nn-1)*T;              % Time vector
NFFT = 2^nextpow2(nn); % Next power of 2 from length of y
Y = fft(L,NFFT);
f = Fs/2*linspace(0,1,NFFT/2+1);
SegmentLength = NFFT;
[P,F] = pwelch(Y,ones(SegmentLength,0),1,NFFT,Fs,'power');
figure,
plot(F,10*log10(P))
title('\bf\itTime waveform for systolic Murmur')
xlabel('Time (msec)')
ylabel('logarithm of the intensity')

figure
Y(f<=300|f>=10000) = 0;

plot(f,2*abs(Y(1:NFFT/2+1)));
hold on
plot(f,2*abs(Y(1:NFFT/2+1)),'k*');
title('\bf\it Line spectrum  for Systolic Murmur for 11 age groupe ')
xlabel('Frequency (Hz)')
ylabel('Amplitude(dB)')

figure
plot(f,2*abs(Y(1:NFFT/2+1)));
hold on
plot(f,2*abs(Y(1:NFFT/2+1)),'k*');
title('\bf\it Line spectrum for Systolic Murmur for 11 age groupe ')
xlabel('Frequency (Hz)')
ylabel('Amplitude(dB)')
figure,
plot(F,10*log10(P))
title('\bf\itPSD of sound for Systolic Murmur for 11 age groupe ')
xlabel('frequency(Hz)')
ylabel('PSD(dB)')
```

Appendix H: MATLAB code for plotting For Systolic murmur (23)

```
close all
clear all
%fid = fopen('cday.wav');
L = audioread('systolic23.wav');
nn = length(L); % Length of signal
Fs = 20000;                  % Sampling frequency
```

```
T = 1/Fs;                    % Sample time
 t = (0:nn-1)*T;               % Time vector
NFFT = 2^nextpow2(nn); % Next power of 2 from length of y
Y = fft(L,NFFT);
f = Fs/2*linspace(0,1,NFFT/2+1);
SegmentLength = NFFT;
[P,F] = pwelch(Y,ones(SegmentLength,0),1,NFFT,Fs,'power');
figure,
Y(f<=300|f>=10000) = 0;
plot(F,10*log10(P))
title('\bf\itTime waveform for systolic Murmur for 5 age groupe ')
xlabel('Time (msec)')
ylabel('logarithm of the intensity')

figure
plot(f,2*abs(Y(1:NFFT/2+1)));
hold on
plot(f,2*abs(Y(1:NFFT/2+1)),'k*');
title('\bf\it Line spectrum  for Systolic Murmur for 5 age groupe ')
xlabel('Frequency (Hz)')
ylabel('Amplitude(dB)')

figure
plot(f,2*abs(Y(1:NFFT/2+1)));
hold on
plot(f,2*abs(Y(1:NFFT/2+1)),'k*');
title('\bf\it Line spectrum for Systolic Murmur for 5 age groupe ')
xlabel('Frequency (Hz)')
ylabel('Amplitude(dB)')
figure,
plot(F,10*log10(P))
title('\bf\itPSD of sound of  Systolic Murmur for 5 age groupe ')
xlabel('frequency(Hz)')
ylabel('PSD(dB)')
```

Appendix I: MATLAB code for plotting For Systolic murmur (8)

```
close all
clear all
%fid = fopen('cday.wav');
L = audioread('systolic8.wav');
nn = length(L); % Length of signal
Fs = 20000;                    % Sampling frequency
T = 1/Fs;                    % Sample time
t = (0:nn-1)*T;               % Time vector
```

```
NFFT = 2^nextpow2(nn); % Next power of 2 from length of y
Y = fft(L,NFFT);
f = Fs/2*linspace(0,1,NFFT/2+1);
SegmentLength = NFFT;
[P,F] = pwelch(Y,ones(SegmentLength,0),1,NFFT,Fs,'power');
figure,
Y(f<=300|f>=10000) = 0;
plot(F,10*log10(P))
title('\bf\itTime waveform for systolic Murmur for 2 age groupe ')
xlabel('Time (msec)')
ylabel('logarithm of the intensity')

figure
plot(f,2*abs(Y(1:NFFT/2+1)));
hold on
plot(f,2*abs(Y(1:NFFT/2+1)),'k*');
title('\bf\it Line spectrum  for Systolic Murmur for age groupe ')
xlabel('Frequency (Hz)')
ylabel('Amplitude(dB)')

figure
plot(f,2*abs(Y(1:NFFT/2+1)));
hold on
plot(f,2*abs(Y(1:NFFT/2+1)),'k*');
title('\bf\it Line spectrum for Systolic Murmur for 2 age groupe ')
xlabel('Frequency (Hz)')
ylabel('Amplitude(dB)')
figure,
plot(F,10*log10(P))
title('\bf\itPSD of sound for Systolic Murmur  for 2 age groupe ')
xlabel('frequency(Hz)')
ylabel('PSD(dB)')
```